A Multiple Access Protocol of Carrier-Sense Multiple Access with Collision Avoidance using Pilot Tone Technique on Passive Optical Networks

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Abstract

In this thesis, a multiple access protocol of Carrier-Sense Multiple Access with Collision Avoidance (CSMA/CA) using Pilot Tone on Passive Optical Network (PON) is proposed.

PON is a tree-and-branched network. The downstream channel is broadcast in nature whereas the upstream channel uses multi-access. One difficulty in upstream multi-access is that data frames sent from one optical network unit (ONU) are only received by the optical line terminal (OLT), but not the other ONUs. Thus, different ONUs may transmit simultaneously and may lead to collision of data at the aggregation point of the remote node (RN).

Multipoint control protocol (MPCP), a medium access control (MAC) sub-layer protocol, proposed by IEEE802.3ah task force provides an efficient solution that allows request-and-grant mechanism to coordinate the multi-access of data. Some suggest to feedback a portion of the upstream optical power at the remote node such that the channel regulation can be done in a distributed manner.

Background materials on the upstream multi-access problem including its possible solution will be discussed in Chapter 1 and 2. In Chapter 3, the proposed multiple access protocol of Carrier-Sense Multiple Access with Collision Avoidance (CSMA/CA) using Pilot Tone will be discussed and its performance will be investigated. Chapter 4 will provide some variations on the protocol to achieve various goals: higher utilization, elimination of capture effect brought by Binary Exponential Backoff algorithm, and enabling class of service (CoS) support.

摘要

本論文提出了一種解決在無源光網絡(PON)中多路接入(Multiple Access)問題的方法，它是利用載波偵聽多路訪問(CSMA)和導頻音(Pilot Tone)以防止沖突(Collision Avoidance)的·

PON 本身是一個樹形網絡，在下行的方向會使用“廣播及選擇”的模式，在上行的方向卻是以多路接入的方式去傳送·在上行的多路接入中，由於所有從光網絡單元(ONU)發出的訊號都是單一地由光線路終端(OLT)接收，所以很難避免有兩個或以上的 ONU 同時上傳資料而做成資料在網絡滙合處沖突·

IEEE802.3ah 工作小組提出了一個以媒體訪問控制層(Medium Access Control)來解決這問題的多點控制協定(Multipoint Control Protocol)，這個用“要求及批准”方法的協定可有效地去防止沖突·也有人提出將一小部分上行光訊號折回到所有的 ONU，這樣所有 ONU 就可以知道其他 ONU 的狀況，沖突問題也因此得以解決·

論文的第一章及第二章將會介紹無源光網絡中的多路接入問題並討論其可能的解決方案·在第三章中，我們會提出一個利用 CSMA 和導頻音去防止沖突的方法，並分析其表現·第四章會在我們提出的方案中再加上一些修改，以求達到更高的輸出率，及實現業務類別(Class of Service)

Contents

Chapter 1:

Introduction

At the beginning of human history, communication is one of the principal needs. From time to time, from place to place, information flows via different media: texts, pictures, language and art. During wartime, the speed of communication is a key factor in deciding who the winner is.

In the 21st century, we are able to communicate "at the speed of light". Wireless communication and satellite communication bring information everywhere through the atmosphere. An important medium was developed in the 20th century- optical fiber. It enables human to transmit large amount of information through a tiny silica tunnel at a speed of $2 \times 10^8 m/s$. Both wireless and optical communications have become very mature in the late 20th century. The task for information engineers of this century is to find out how to make use of these technologies and interchange information boundlessly over the world in high speed and with enormous amount.

1.1. First Mile Evolution

Some old texts refer "the first mile" as "the last mile". It has been renamed to "the first mile" by networking community to emphasize its priority and importance [1]. The first mile network segment connects the service providers' central offices to business and residential subscribers. Subscribers demand first-mile access solution as it is broadband, offering Internet media-rich services and is reasonable in price with existing networks. In the old days (10 years ago), network operators provides first-mile access over Public Switched Telephone Network (PSTN). PSTN has been a long proven stable network for voice transmission. However, there are different requirements for voice and data service. Traditional PSTN can only provide 56 kbps of data service, which could not satisfy the need for media-rich content such as picture, movie and high-quality music. Telecom engineers started to develop new cost-effective technology to provide broadband access to residential subscribers.

Various technologies have been developed to provide broadband service to commercial and residential subscribers. Incumbent telephone companies have deployed residential broadband service by overlaying Digital Subscriber Line (DSL) technology on their existing telephone networks [2]. DSL uses the same twisted pair as telephone lines and only requires a DSL modem at the customer sides. Digital Subscriber Line Access Multiplexer (DSLAM) is use to multiplex the broadband signal in the central office (CO). The downstream data rate provided by ADSL ranges from 128 Kbps to 6 Mbps depending on the distance between the subscriber and the central office, while upstream data rate ranges from 128 Kbps to 640 Kbps. Hence, the ADSL traffic is asymmetric.

Cable television (CATV) companies try to provide broadband service by integrating data services over their coaxial cable networks, which were originally designed for analog video broadcast [3]. CATV service provider uses hybrid fiber coax (HFC) networks by having optical fiber between the head-end at central office and an optical drop at subscriber end. The signal at subscriber end was dropped into coaxial cable, repeaters and tap couplers. Finally, it was demultiplexed, demodulated and decoded. The drawback of overlaying data services over HFC networks is that each shared optical node has to be shared among some tens to hundreds of subscribers. Hence during peak hours, subscribers may experience slow speed of data transmission.

New companies start to provide broadband service by bringing fiber to the building (FTTB) and fiber to the home (FTTH). Different from telephone companies and cable TV companies, fiber is used to feed the data into the subscriber homes or buildings. This shortens the length of copper cable used which limits the speed. These newly deployed networks are capable to provide data speeds of gigabit per second and costs comparable to those of DSL and HFC networks.

1.2. Access: Passive Optical Network (PON)

Optical fiber is the most suitable candidate to deliver bandwidth-intensive integrated voice, data and video services over 20 km in access network. A reasonable way to deploy optical fiber is using an optical fiber to connect each subscriber and central offices. This is known as point-to-point topology (P2P). With this simple topology, it is cost prohibitive because it requires outside plant fiber deployment. Considering N subscribers at an average distance L km from the central office (CO), P2P topology requires $2N$ transceivers, N on central office side and N on subscriber side. Fiber use is NL km in total length from CO to outside plant. (Assuming negligible distance between outside plant and subscribers)

To reduce the length of fiber, curb switch is used. Similar to HFC case, an optical fiber is connecting the central office and the curb switch. N fibers are used to connect the switch and subscribers. This greatly reduces the fiber length N times. However, two more transceivers are needed for the communication between CO and switch.

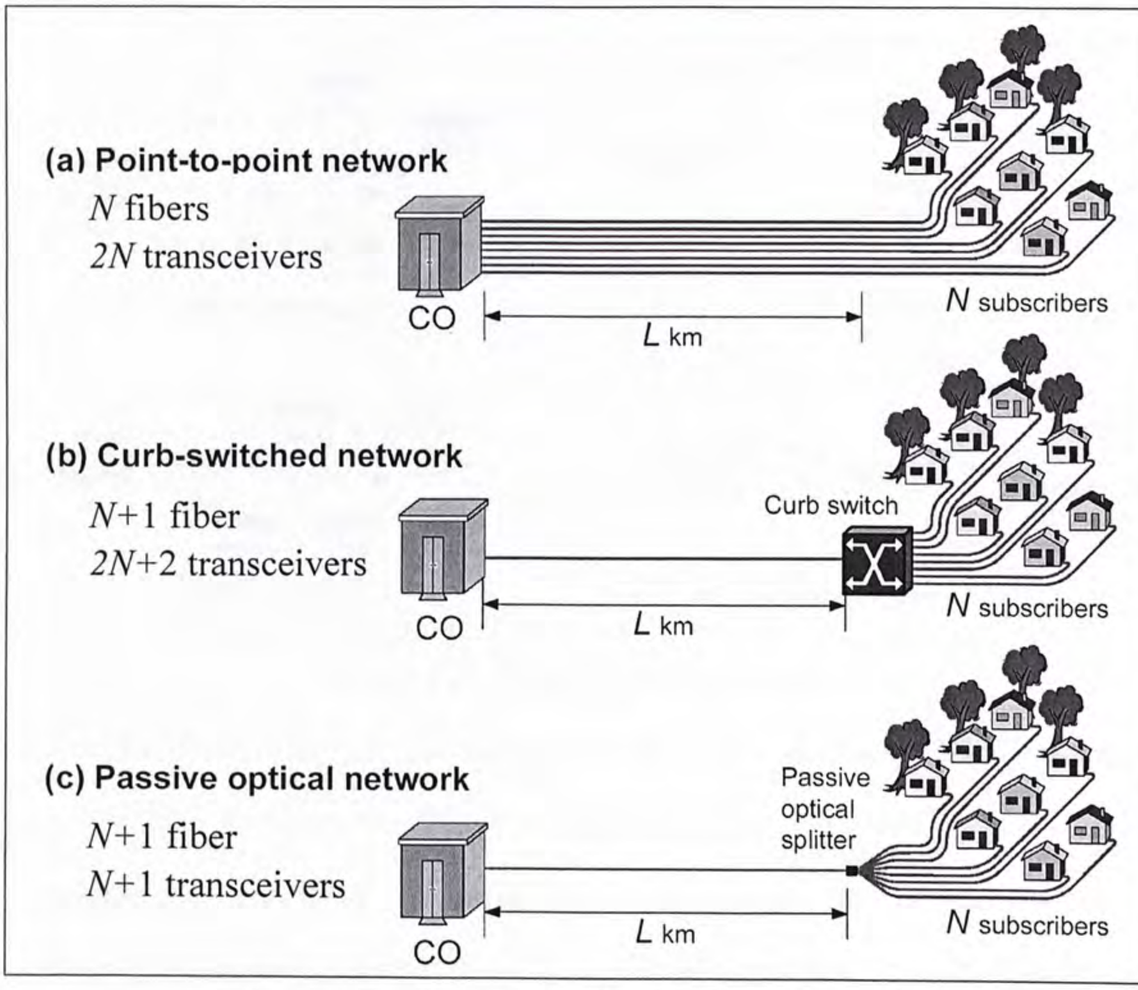

Figure 1.1 Three Types of Optical Access Network

Hence it is reasonable to replace the expensive curb switches with inexpensive passive optical splitters. Passive optical networks (PONs) are point-to-multipoint (P2MP) fiber optical networks. A single, shared optical fiber of a PON connects the central office to an optical splitter located near customers. In such way, each customer receives a dedicated optical fiber but shares the long distribution trunk fiber.

High speed communication is made possible at the customer end with low cost intermediate device. The word "passive" describes that there is no electrical power input to any component of the network other than the two endpoints.

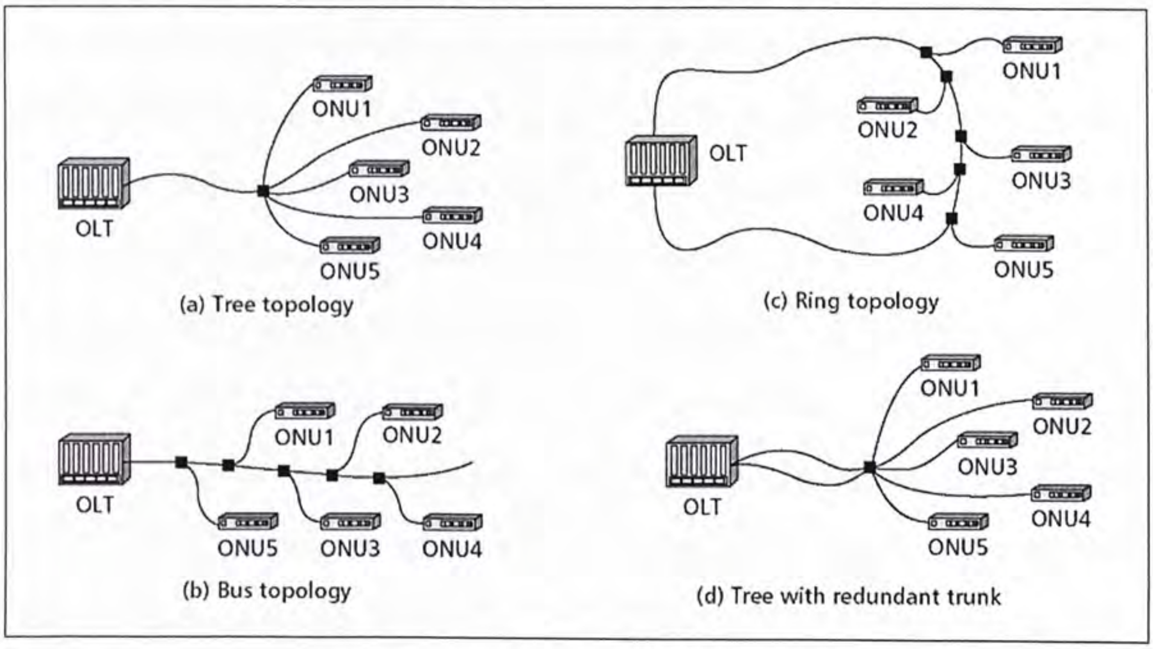

Figure 1.2 Typical PON Architecture

Figure 1.2 shows some typical configurations for PON. A single PON can usually supports up to 32 customers. Network engineers deploy them into a tree-and-branch topology, by using either *1:2* asymmetrical optical tap couplers or *1:N* optical couplers as illustrate in figure 1.2b and 1.2a. Protective path can be added as shown in figure 1.2c and 1.2d. Operators can move optical splitters closer to or further away from the local central office, without affecting active end terminals. In this way, they can optimize network deployment costs.

An advantage of using P2MP networks over multiple P2P networks is that P2MP networks requires less amount of fibers and optical transceivers deployed inside the central office and outside cable plant. PON does not need electrically powered box in the field, when comparing with a hybrid fiber coaxial network (HFC). Its optically transparent nature makes it upgradeable by increasing channel bit rate or by adding additional wavelengths. However, what is challenging network engineer is that tree-and-branch topology requires a more complex media access communication protocol than that of a full-duplex P2P network. In addition, as PON is a passive network technology, the network elements do not amplify the signal. Therefore, trunk lengths and the number of signal splits are limited.

PON network elements consist of Optical Line Terminals (OLTs), Optical Network Terminals (ONTs), Optical Network Units (ONUs), and passive splitters. The OLT can either generate optical signals on its own, or pass optical signals (e.g., SONET or DWDM) from a collocated optical cross-connect or other device, broadcasting them downstream through one or more ports. The ONU or ONT terminates the circuit at the far end.

There are two standards for PON up to the present time. One is APON with the use of ATM protocol as the data format inside the fiber. The other one is EPON with the use of Ethernet protocol.

1.2.1. ATM-PON (APON)

APON (ATM-based Passive Optical Network) runs the ATM protocol, which is favored by the ILECs for DSL and for their internal backbone networks. So, ADSL and G.lite can run nicely over APON in a hybrid fiber/copper network. Unlike Ethernet packet, ATM cell has fixed length. Therefore, synchronization is easier for both upstream and downstream traffic. Full Service Access Network (FSAN) is the group which proposed the use of APON [9].

FSAN group was created by a group of service providers in order to facilitate the creation of suitable access network equipment standards and hence reduce the price of equipment. The implementation proposed in ITU G.983 uses ATM protocol.

Asymmetric traffic was used, running at speeds of 622 Mbps downstream over a wavelength between 1480 nm and 1580 nm and 155 Mbps upstream between 1260 nm and 1360 nm. Upstream and downstream traffic can be transmitted over separate fibers, or can share a single fiber through WDM (Wavelength Division Multiplexing). Upstream signals are supported by a time-division multiple access scheme, with the transmitters in the ONUs operating in burst mode. FSAN supports both symmetric and asymmetric modes.

FSAN trunk lengths can be up to 12 miles, and as many as 32 users and 64 endpoints can be supported per trunk at the current speeds and with the current splitter technologies. Currently, the group is developing a new implementation GPON derived from their old version BPON. GPON supports upstream traffics from 155 Mbps to 2.5 Gbps, while downstream traffics can go from 1.25 Gbps to 2.5 Gbps. There are also working groups in FSAN on the optical network maintenance and provisioning, which is another important issue in PON technology.

1.2.2. Ethernet PON (EPON)

In 1998, IEEE announced the Gigabit Ethernet (GigE) standard. In December of 2000, IEEE802.3 established the IEEE802.3ah Ethernet in the First Mile (EFM) group to study the use of Ethernet on access network [11]. Their studies included Ethernet on VDSL (EoVDSL), on point-to-point optical fiber connection and on PON (EPON). In addition, the group also defined how Ethernet protocol works with network operation and maintenance activities (OAM) to make the network more robust.

Another important group in EPON development is Ethernet in the First Mile Alliance (EFMA) [10]. It is an industry alliance which supports the standardization process with resources. EFMA has four main targets:

- ✓ Support the Ethernet in the First Mile standards effort conducted in the IEEE 802.3ah Task Force;
- ✓ Contribute technical resources to facilitate convergence and consensus on technical specifications;
- ✓ Promote industry awareness, acceptance, and advancement of the Ethernet in the First Mile standard and products;
- ✓ Provide resources to establish and demonstrate multi-vendor interoperability and encourage and promote interoperability events.

EPON standard is built on a medium access control protocol, Multi-point control protocol (MPCP). MPCP uses message, state machine and timer to control access to P2MP structure. The systems that implement MPCP only allow one ONU send data to OLT at a time. The OLT is responsible for congestion control, bandwidth allocation, synchronization and traffic report. The protocol also supports automatic registration to make the management easier. Currently, EFMA aims at providing EPON services with speed 1.25 Gbps over 20 km.

IP/Ethernet covers over 95% of local area network (LAN) in use. With the development of 10-Gigabit Ethernet (10GigE), EPON is a good choice to solve the first mile problem. The problem remaining to telecommunication engineers is to feed the low bit rate LAN Ethernet signal to high speed EPON.

1.3. Problem Definition and Possible Solutions

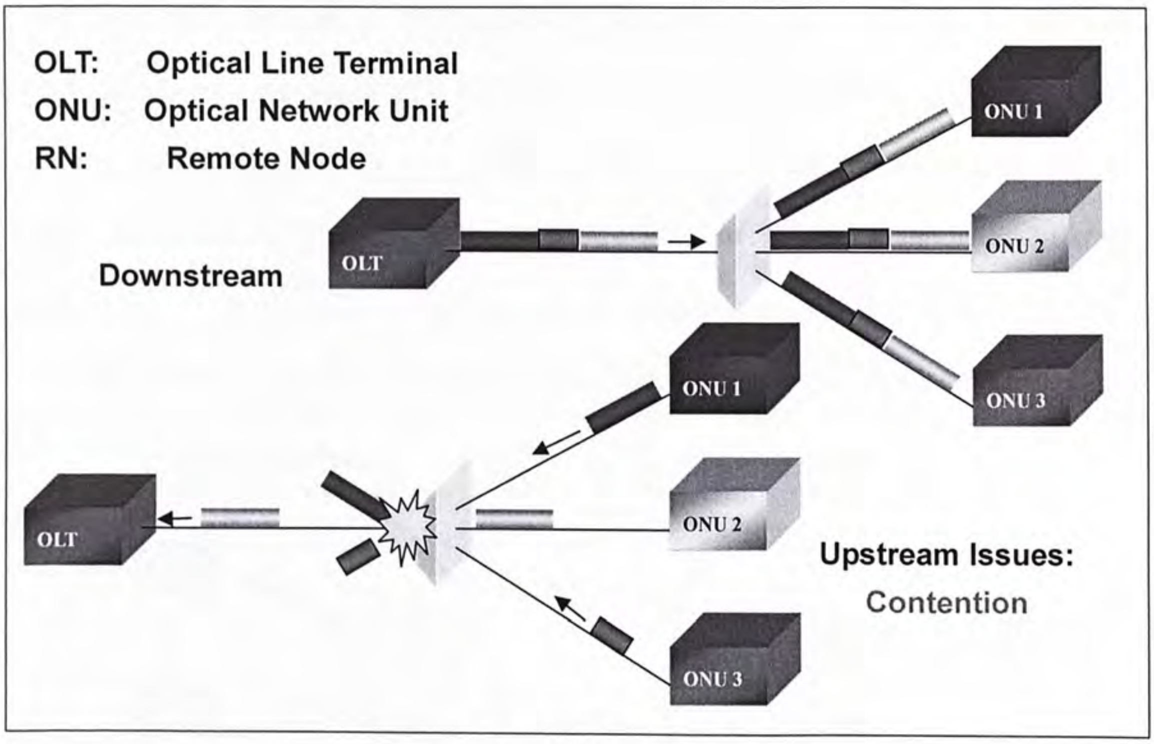

Figure 1.3 Problem Definition

In tree-and-branch network, PON in our case, the downstream channel is broadcast in nature while the upstream channel uses multi-access. One difficulty in upstream multi-access is that data frames sent from one ONU are only received by the optical line terminal (OLT), but not the other ONUs. Thus, different ONUs may transmit simultaneously and may lead to collision of data at the aggregation point of the remote node (RN). There are several ways to solve this problem, in different domain.

1.3.1. Wavelength Division Multiplexing (WDM)

The development of WDM technology allows us to increase the capacity of fiber trunks more than 100 times by means of DWDM. One possible way to separate the ONU upstream channels in PON upstream problem is to use WDM by giving each ONU a dedicated wavelength [4, 5]. Some designs of WDM PON replace the power splitter in original point by wavelength splitting and combining equipment such as array waveguide grating (AWG). As this kind of WDM PON uses different equipment, it is common to distinguish it from power splitting PON (PSPON). Tunable receiver or receiver array at OLT is needed to receive multiple channels.

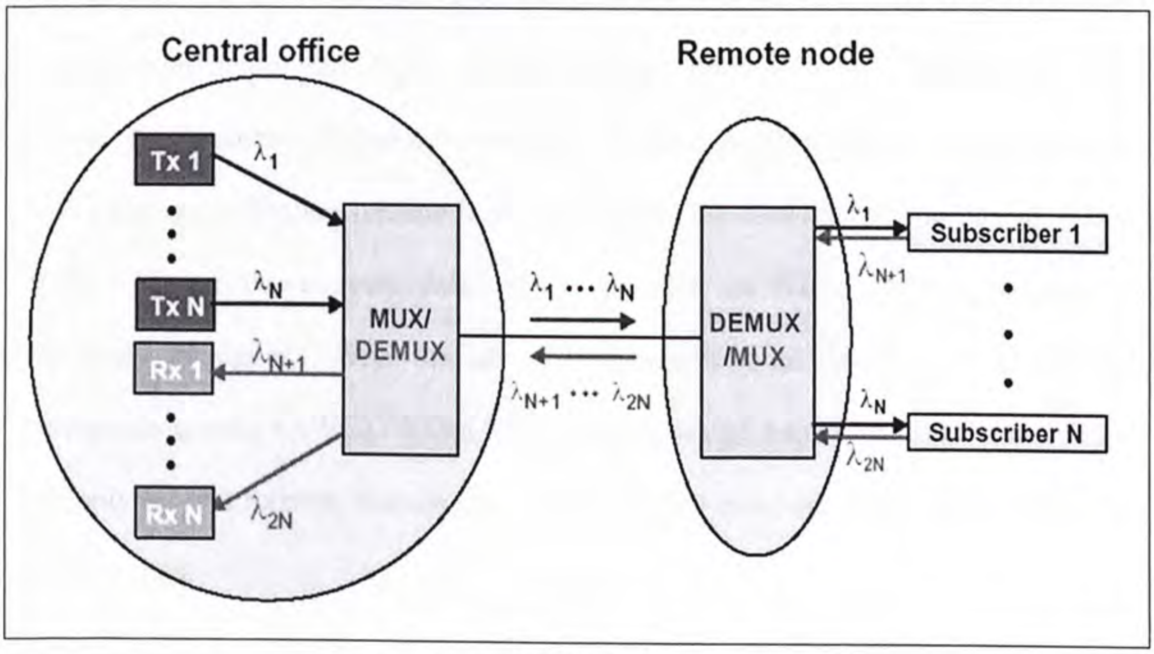

Figure 1.4 WDM PON Architecture

In WDM PON, there is no upstream collision problem as every ONU uses different upstream wavelength. It is a simple solution to the problem. At the same time, it remains cost-prohibitive for access network. The WDM solution requires either a tunable receiver, or a receiver array at the OLT to receive multiple channels. Furthermore, wavelength-specific ONU inventory is a more serious problem for network operators. Each ONU is packed with a laser with narrow and controlled spectral width, making the cost for ONU more expensive. Using tunable laser is a solution, but it is still too expensive for the use on access networks. Hence, WDM PON is not attractive to network operators at this moment.

There are several variations on WDM PON. One variation is to remodulate the downstream wavelength. In this solution, the ONU modulates its upstream signal onto the downstream wavelength directed to it. This solves the wavelength-specific problem for ONU. But the cost still remains high as external modulator and amplifiers are being used in ONU instead of direct modulation in TDM PON.

Recent development of coarse wavelength division multiplexing (CWDM), passive AWG and cost-effective tunable laser has become the enabling technology of WDM PON. Higher downstream data rate is possible in WDM PON as bandwidth efficiency is higher. With the use of wavelength route device such as arrayed waveguide grating (AWG), WDM PON is more secure than TDM PON as an ONU can only receive its own wavelength. However, the cost issue is still a barrier for the development.

1.3.2. Time Division Multiplexing (TDM)

Time Division Multiplexing (TDM) is widely used in traditional networks such as SONET ring, Ethernet, PSTN and GSM networks. Different access unit occupies the channel at different times. If all units followed the rule defined by the protocol, there should not be two active units sending data at the same time, or collision will be detected and then collided data will be retransmitted. In PON, TDM approach has the greatest number of supporters, for instance, APON and EPON uses TDMA. According to the time dividing strategy, there are four common types:

- ✓ Fully synchronized: Ranging protocol is used so that propagation time from each ONU is known. If there are *N* ONUs on the network, the channel will be divided into *N* parts in time domain. Every ONU can only send data to the OLT at its own assigned time slot. Capacity is shared in this case. If there is only one active ONU, the utilization is *1/N*. Besides from low utilization, synchronization is rather complex to implement.
- ✓ Contention: Each ONU transmits when ready. The OLT will acknowledge correctly received packets. If there is collision, involved units will retransmit collided packets. Utilization around 18% can be achieved in pure Aloha scheme where higher utilization is possible by the use of CSMA type scheme. The advantage of this strategy is that it is easy to implement. In PON, contention-based solution can be made by either notifying collision by OLT or feedback some upstream optical power to all ONUs. However, the large delay and low utilization make it unfavorable for high speed applications.
- ✓ OLT Polls: The OLT will poll its ONUs for data transmission. Each ONU will only transmit after it is being polled. Usually, MAC layer protocol is used as polling and control message. As there is no waste of time slot for collision and resent packet, the channel utilization can be high (>90%). In both APON and EPON, upstream collision is avoided by polling and MAC protocol.

1.3.3. Sub-carrier Multiplexing (SCM) & Frequency Division Multiplexing (FDM)

Except multiplexing in time domain and optical domain, we can also multiplex the signal in electrical domain. Sub-carrier multiplexing has been used in ADSL systems. Different ONU is assigned with a different carrier frequency. At the OLT, sub-carrier demultiplexing is performed. This solution is not suitable in PON environment because near-far effect, shot noise and optical interference limit the performance. Its low-bit rate and low cost efficiency are also unfavorable factors.

1.3.4. Code Division Multi Access (CDMA)

Code Division Multi Access has shown its importance in 3G cellular networks. In a CDMA PON, each customer's channel is given its own code for spreading and dispreading. Ref. [8] claims that CDMA PON is more cost effective and simpler. It is transparent to input channel's data protocol. The optical beat noise problem [6, 7], which appears in SCM approach, does not have much effect on the CDMA-PON system. One disadvantage was found when comparing the channel utilization, CDMA-PON has lower channel utilization than TDM PON.

1.4. Thesis Organization

This chapter gives a general picture on access networks. Two types of PON, ATM-PON and Ethernet PON, are studied. After bringing out the P2MP upstream access problem, we have studied and discussed some possible solutions. Finally, we conclude that TDM approach is the most suitable candidate for multiple-access on PON.

Chapter 2 gives the motivation on proposing the CSMA/CA using pilot tone protocol. We will study two current solutions, MPCP of EPON and CSMA/CD. MPCP provides a centralized solution where all scheduling works are done in the OLT. We will also discuss solutions using distributed approach where scheduling is done by all active ONUs on the PON.

With the inspiration from MPCP and CSMA/CD solution, solution that takes the advantage of pilot tone will be proposed in Chapter 3. The aim of this solution is to avoid data collision by carrier-sensing. Similar to MPCP, collision is avoided by ONU making request before sending packet. Hence the solution is named "CSMA/CA using Pilot Tone". Simulation works have been done using JAVA SDK. Result will be shown and analyzed in this chapter.

In Chapter 4, we will try to solve some possible problems that may occur on the proposed protocol. The first problem we have resolved is the low channel utilization when small packet size is used. Modification is made based on sending more packets per request. Then, we will propose a solution for the fairness problem brought by capture effect under heavy loading. Lastly, by using the same technique, we will show how to implement priority on the network, which enables the implement of CoS on the protocol possible.

Chapter 2:

Background

In the previous chapter, we have discussed the need of solving the upstream multi-access problem. WDM-PON provides a solution with high utilization. But the cost for tunable lasers, receiver array and WDM components is still too high for the access application. CDMA PON cannot give high channel utilization. Hence, TDM approach seems to be the most suitable solution for this problem. MPCP, a media access control (MAC) sub-layer protocol, proposed by IEEE802.3ah task force provides an efficient solution that allows request-and-grant mechanism to coordinate the multi-access of data [11]. Nearly 95% channel utilization and quality of service (QoS) can be achieved by controlling the granted data size and priority.

In this chapter, we will first review two approaches that resolve the problem. Later in this chapter, we will present the motivation on solving the problem using pilot tone technique.

2.1. EPON Solution: - MPCP

EPON aims at using Ethernet as layer-2 protocol on PON. Hence there is no need for data format conversion between ONU, network and OLT. IEEE 802.3 standard has defined two basic configurations for Ethernet. One configuration is to use CSMA/CD protocol over the shared medium. Another configuration is to connect stations through a switch using full-duplex P2P links. However, in PON, it cannot be simply considered as either a shared medium or a point-to-point network.

IEEE802.3ah proposed the use of a layer-2 non-contention solution known as multipoint control protocol (MPCP). The general idea of this scheme is to synchronize all ONUs to a common time reference. ONU should only send packet at its own timeslot. Each timeslot is capable to carry several Ethernet packets. All ONUs will buffer the received packets from its clients and burst all stored packets at channel speed when its timeslot arrives. If the buffer is empty when its timeslot arrives, a 10-bit characters indicating idle are transmitted. Timeslot allocation can be either static (fully synchronized as described in chapter 1) or dynamic. Different bandwidth allocation scheme can be employed in MPCP as MPCP is a supporting protocol rather than a standard.

The timeslot allocation process can be either centralized in the OLT or distributed among the ONUs. The distributed approach has a fundamental limitation: all ONUs should be fully connected. This is not desirable as it requires more fiber deployment. MPCP has chosen the centralized approach.

MPCP relies on two Ethernet messages: GATE and REPORT. GATE message is sent from the OLT to an ONU on the downstream channel to assign a transmission timeslot. REPORT message is used by an ONU to tell the OLT about its conditions such as buffer status. OLT has the intelligent unit to make the allocation decision. Both GATE and REPORT messages are MAC control frames of type 88-80 and are processed by the MAC control sub-layer of ONUs and OLT. Figure 2.1 and 2.2 show the operation of GATE and REPORT message.

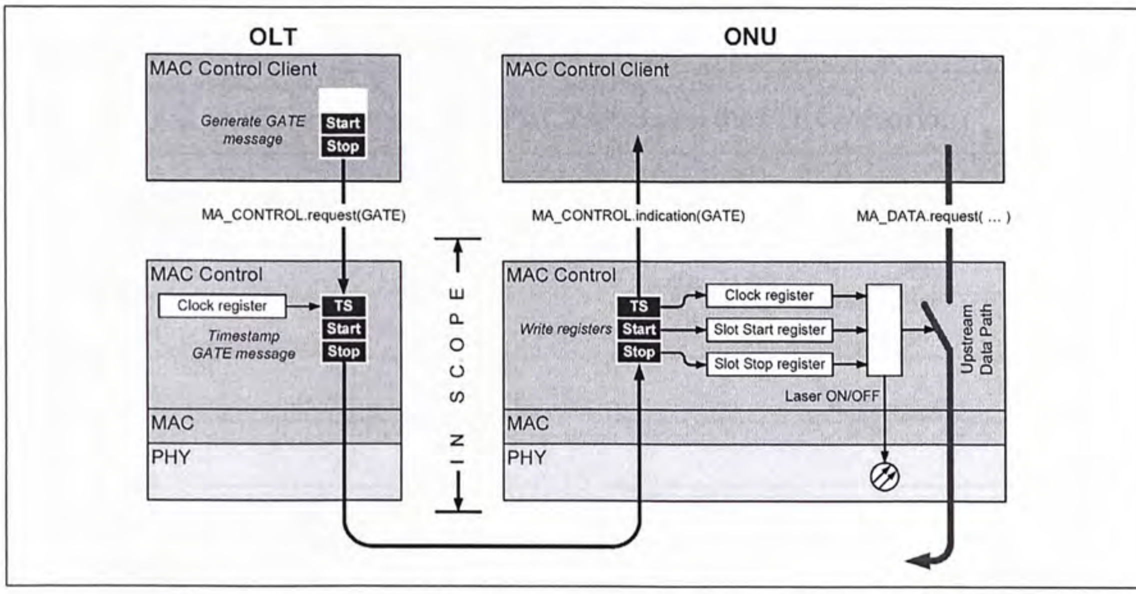

Figure 2.1 Sending GATE Message with MPCP [IEEE 802.3ah]

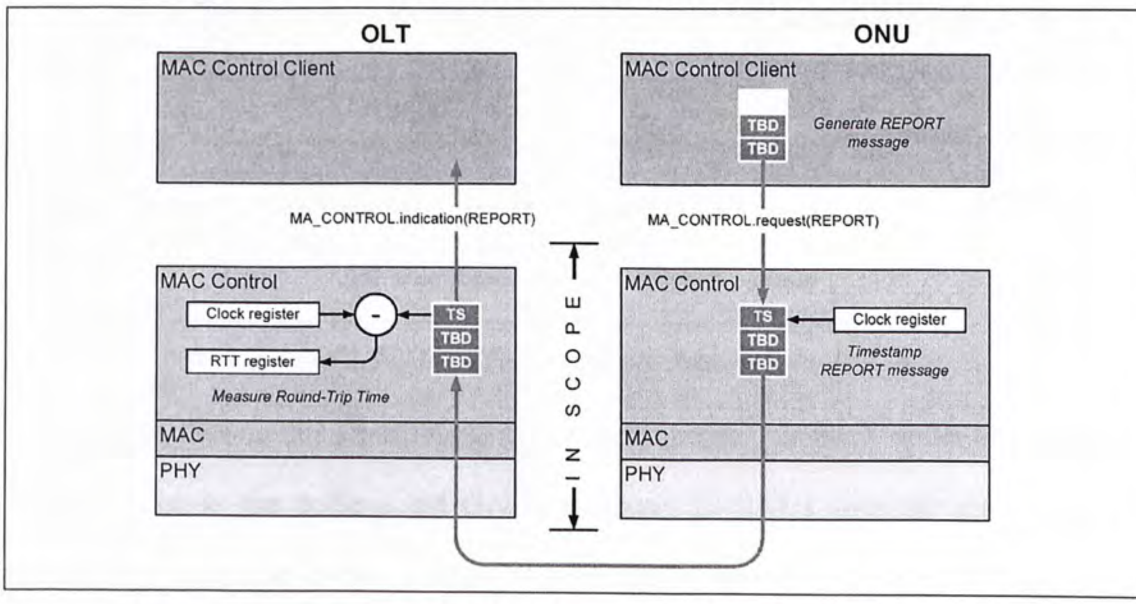

Figure 2.2 Sending REPORT Message with MPCP [IEEE 802.3ah]

In MPCP, ONU initialization and knowledge of round-trip delay of each ONU are needed. ONU initialization is used to detect newly connected ONUs and learn the round-trip delay of that ONU. After the ONU is initialized, it is synchronized with OLT and polling will be carried to all initialized ONU.

Under normal operation, MPCP is like a polling algorithm. In [12], Alloptic, a member of EFMA, have proposed and studied an IPACT protocol based on MPCP. Below shows an illustration on how IPACT works on the PON scenario:

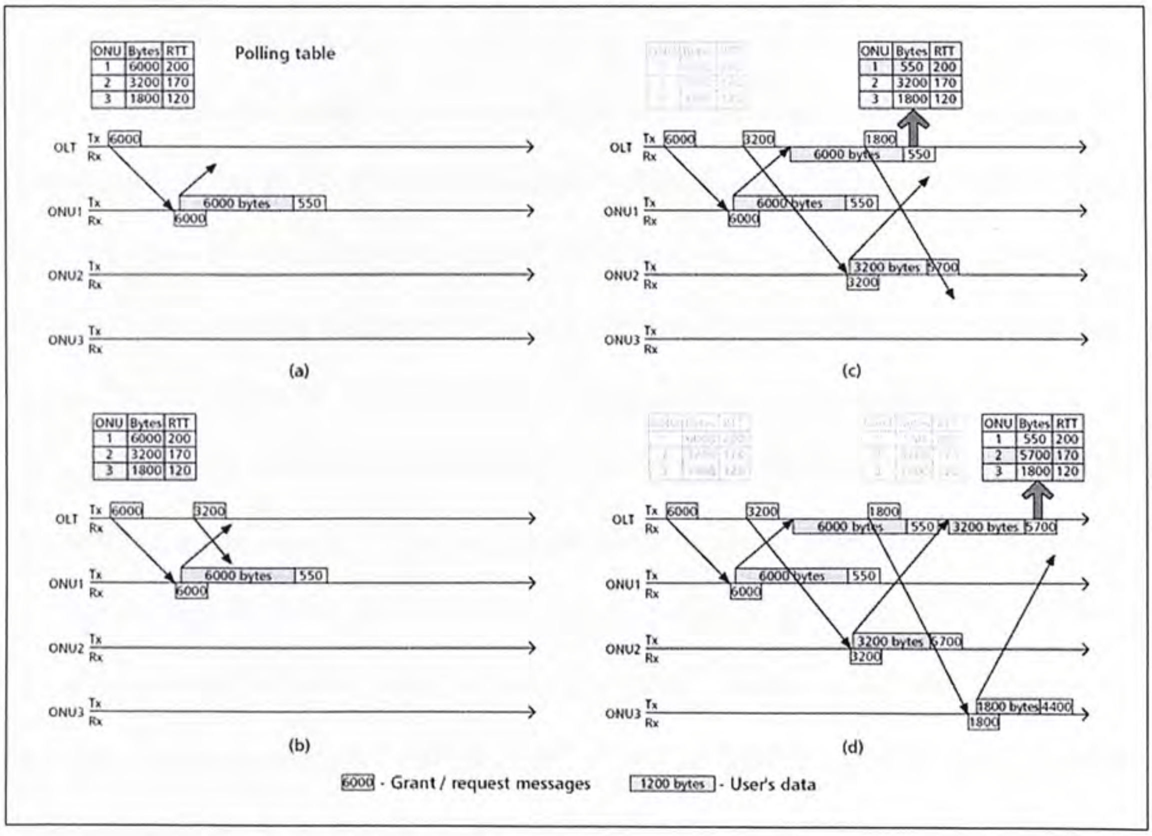

Figure 2.3 IPACT: A Dynamic Protocol for EPON

In depth studies on this algorithm is shown in the paper. In brief, under this example, the OLT sends the polling and GATE messages to ONUs with the knowledge of round-trip time and buffer status of each ONU. Round-trip delay is taken into consider when ONUs transmit frames.

Several bandwidth allocation schemes were studied and stimulated in [12]. Bandwidth allocation is done by controlling the grant window size (number of bytes allow to be sent by ONU).

2.2. CSMA/CD on PON

Carrier sense multiple access with collision detection (CSMA/CD) scheme has been used as an effective multiple access control for Ethernet built with bus topology for many years. On PON, CSMA/CD cannot be implemented because data from each ONU is directly transmitted to the OLT without passing through its neighboring ONUs. Hence when multiple ONUs send data at the same time, the collided data cannot be read at the OLT and collision is not known by the involved ONUs.

A possible solution is to detect collision at the OLT and then the OLT informs ONUs by sending a jam signal. However, propagation delays in PON, typically with over 20 km in length, can greatly reduce the efficiency of this scheme. In addition, contention-based schemes have a drawback that it cannot provide a deterministic service. Hence, quality of service (QoS) cannot be implementing on such systems. Access networks must support integrated voice and video services, in additional to data. Hence the network must provide some guarantees to such time-sensitive traffics.

With a long history of success of Ethernet, CSMA/CD and Ethernet seems to be inseparable. Therefore, engineers are seeking ways to implement CSMA/CD on EPON for multiple access control. In a recent multiple access research on PON, it is proposed to use optical CSMA/CD [13] by feeding back a small part of the upstream optical power to all ONUs such that every ONU knows the status of the channel. This is a distributed bandwidth-allocation approach, and there is no need for synchronization and distance concern. Figure 2.4 shows the schematics of the design.

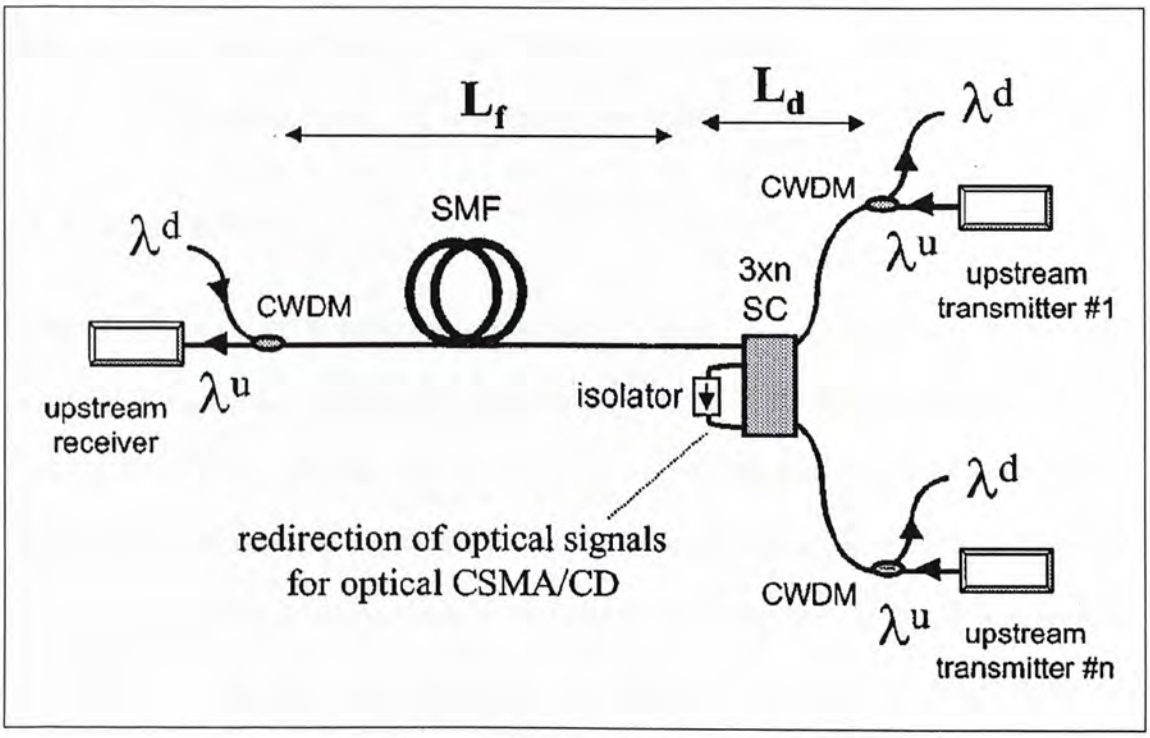

Figure 2.4 Schematic of CSMA/CD over EPON [13]

In this scheme, the upstream and downstream packets are transported over different wavelengths. Coarse WDM (CWDM) multiplexer and demultiplexer are used to separate these wavelengths. Downstream transport can be based on the well developed broadcast and select protocol. Each ONU uses the same upstream wavelength. At remote node (RN), some optical power is fed back from the 3xN star coupler (SC). The feedback signal is received at all ONUs and then passes through a low-pass filter and a decision circuit to identify collision if any.

Since this scheme will feedback the upstream data and "broadcast" it to all ONUs, it is considered not secure. Hence in [14], the author proposed to use loop of different length at remote node to "self-jam" the reflected signal while keeping the signal to OLT clean.

The CSMA/CD approach provides a solution for the problem. Synchronization and knowledge of round trip time of each ONU are not needed. However, the collision causes wasting of timeslots. The consequence is the low channel utilization.

2.3. Motivation

The EPON and APON approach can effectively solve the problem and provide high channel utilization. CSMA/CD approach provides a simple architecture for the multiple accesses. In this thesis, we try to find a way that has the advantages of these two approaches at the same time. We will introduce collision avoidance into CSMA/CD approach as we believe the channel utilization will be higher without collision. To do this, extra resources are needed for signaling to avoid collision. Pilot tone technique that was used in HORNET [15], which superimposes an RF signal onto original data, is chosen for this purpose. Previous works have successfully shown the use of pilot tone technique for management. It is the first time that using pilot tone technique for upstream access management in PON is proposed.

Collision avoidance is done by request and grant process, as in EPON. In the case of EPON, when an ONU wants to send data, it first tells the OLT its status. The OLT will assign a time slot for this ONU to send its data. In our case, the request and grant process is done by pilot tone. Chapter 3 will give an overview on the proposed protocol.

Chapter 3:

CSMA/CA Protocol using pilot tone on PON

In PON, the downstream channel is broadcast in nature while the upstream channel uses multi-access. One of the difficulties in upstream multi-access is that data packets sent from one ONU are only received by the optical line terminal (OLT), but not the other ONUs. Thus, different ONUs may transmit simultaneously and may eventually lead to collision of data at the aggregation point of the remote node (RN). Chapter 1 has given an overview on current solution to this problem. This chapter will further give an overview on the carrier-sense multiple access protocol with collision avoidance (CSMA/CA) using pilot tone.

3.1. Basic Protocol Description

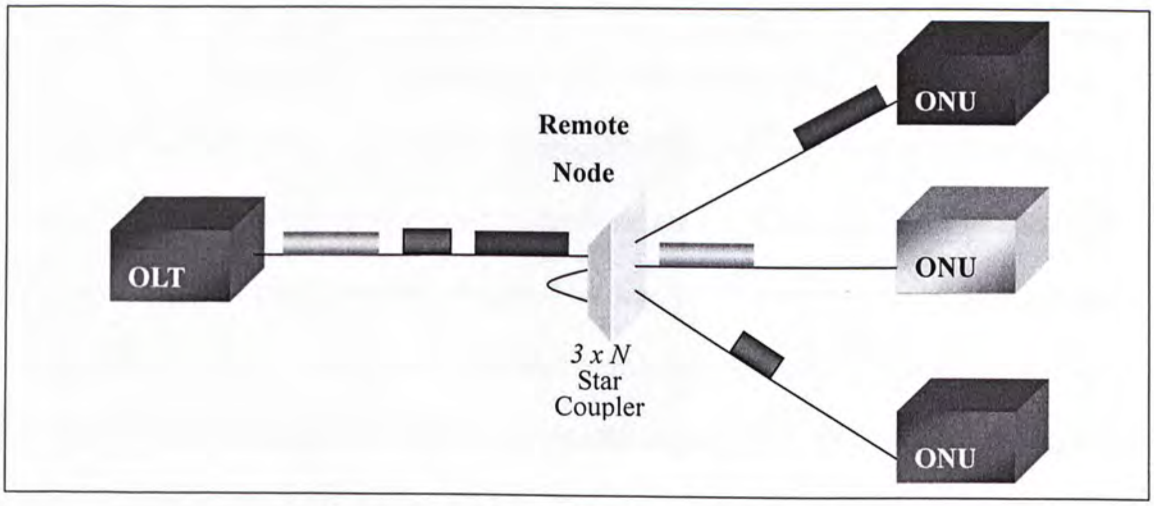

Figure 3.1 Physical Network Connection

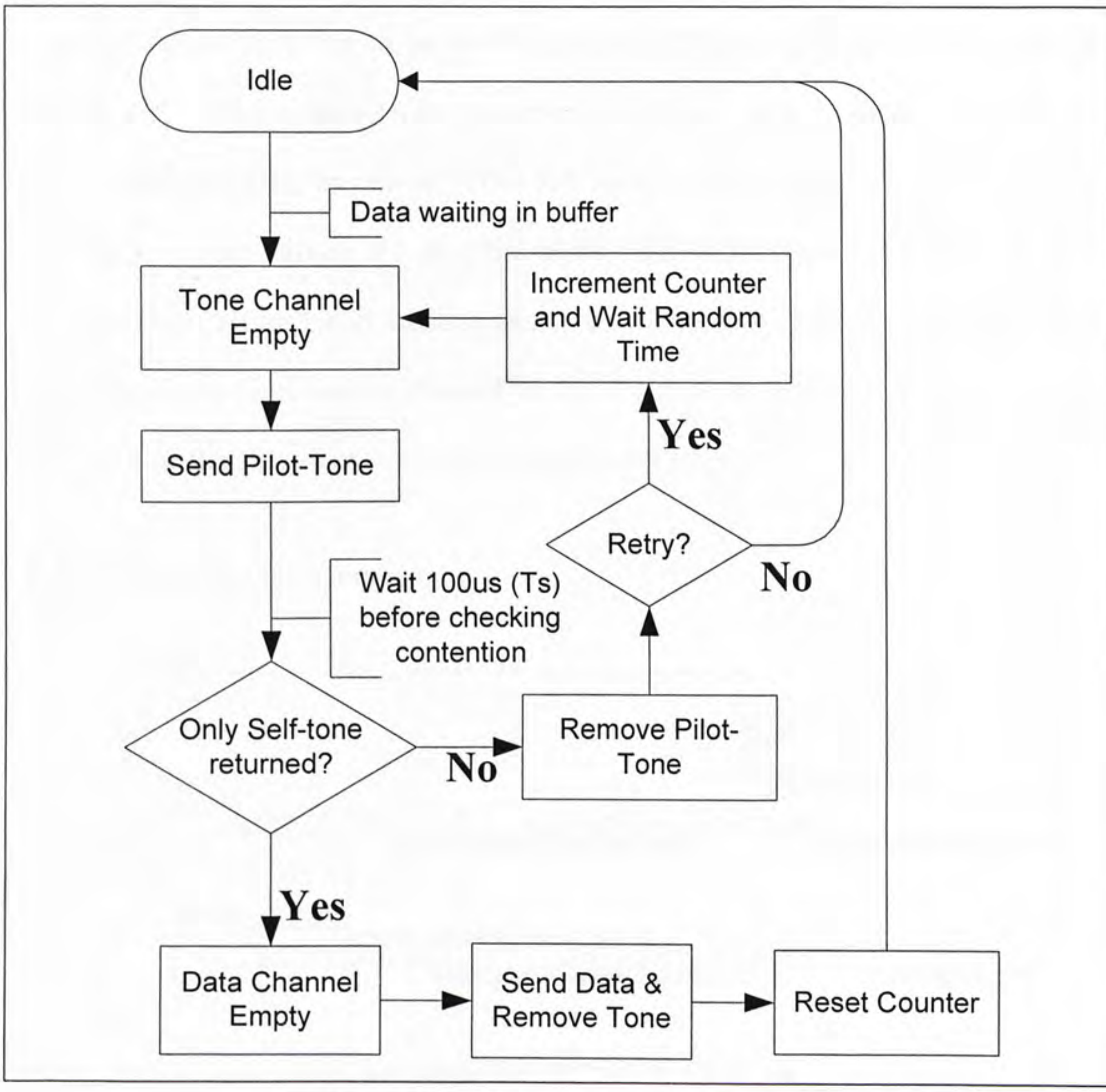

Figure 3.2 CSMA/CA protocol using Pilot Tone on PON

In our proposed scheme, each ONU is assigned with a distinct pilot tone frequency for signaling, which will be multiplexed to the base band data as in [18]. The spectra for the data and the pilot tone are regarded as the data channel and the tone channel, respectively. ONUs send pilot tone before they transmit data. The upstream signal is fed back at the aggregation point of the RN to every ONU. Whenever there are two or more ONUs sending the pilot tone at the same time, all ONUs will sense there is more than one pilot tone in the feedback signal. Based on this feedback signaling, a proposed collision avoidance algorithm will then be performed. Figure 3.2 presents the state diagram which will be carried out when an ONU sends a data packet. In order to assure fairness for all ONU nodes with different distances from the RN, the algorithm "sleeps" after sending pilot tone. The sleep time, T_s, should be set at least larger than the round-trip time of the ONU that is the farthest from the RN. We will see how the algorithm works in following examples.

3.1.1. With No Contention

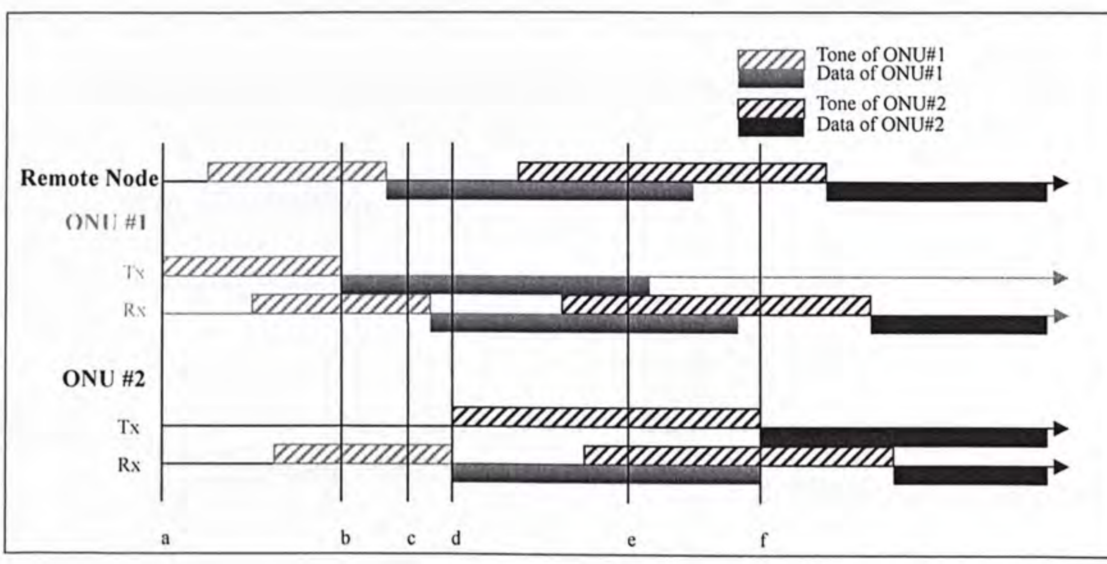

Figure 3.3 Case 1: No Contention

Figure 3.3 shows an example of tone sending and data transmission when there is no contention. In the example, ONU #2 is further away from the congregation point than ONU #1. At time <a>, ONU #1 detects tone channel is empty and therefore it can sends its tone for requesting. Then, it sleeps until time <b> when it senses only its tone present in the feedback tone channel. The algorithm checks the data channel and knows that no ONU is sending data at this moment. Therefore, ONU #1 sends data immediately and removes its tone on the channel at time <b>. At time <c>, data arrives at ONU #2. The algorithm in ONU#2 turns from idle state to the state that checks the tone channel. It finds that the tone channel is not empty. Hence it waits until time <d> when tone channel is cleared, and then it sends the requesting tone. ONU #2 awakes at time <e> and discovers that the tone channel contains only its tone, indicating that it can send data immediately when the data channel is empty. It holds its tone until <f> when it detects ONU#1 has finished sending data. Then, it removes its tone on the tone channel and transmits data. Note that in this example, <b>-<a>=<e>-<d>=T_s. Chapter 4 will discuss some variations of the protocol by having different value of T_s in different ONU to enable priority among ONUs.

3.1.2. With Contention

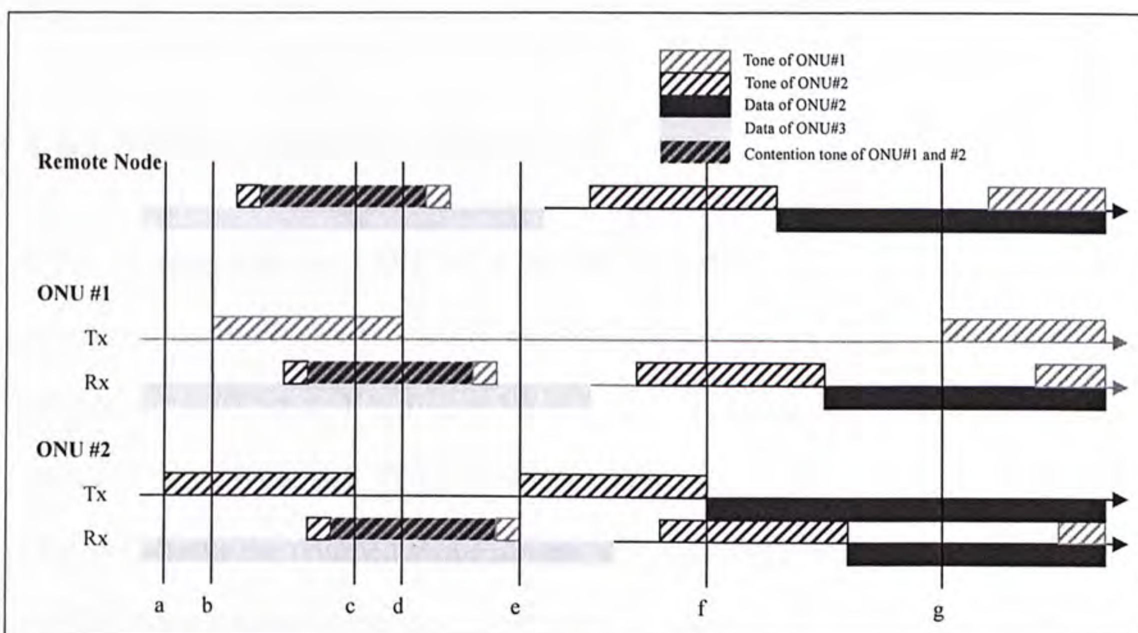

Figure 3.4 Case 2: With Unsolved Contention

Figure 3.4 illustrates the case when there is contention. At time <a>, ONU #2 senses that the tone channel is empty, and then it sends its tone to notify other ONUs. Assuming a brief moment later at <b>, ONU #1 also finds the tone channel is empty, and sends its tone for requesting. The tones of ONU #1 and ONU #2 meet at the aggregation point of the RN and are fed back to all ONUs. ONU #1 and #2 detect that there are more than one tone on the tone channel at time <c> and <d> respectively, and thus both of them remove their tones from the tone channel. According to the algorithm, when contention occurs, both ONUs will retry the requesting after a random back-off time. In this example, ONU #2 has zero back-off time and sends the requesting tone once the tone channel is cleared at time <e>. ONU #1 has a longer back-off time and thus has to wait for a longer time before retry. At time <f>, ONU #2 finds no contention and the data channel is empty, thus it transmits data and removes the tone at the same time. ONU #1 retries at time <g> and the whole process continues.

3.1.3. With Contention and Winner

If the traveling time from ONU#2 to ONU#1 is longer than $0.5T_s$, it is possible that contention occurs between ONU#1 and ONU#2 but one of them cannot detect the contention, as illustrated in figure 3.5 and 3.6. Both cases do not introduce data collision. In Figure 3.5, ONU #1 and ONU #2 send tone at time <a> and <b>. ONU #2 checks the returned signal at time <c> while ONU #1 checks it at time <d>. ONU #2 cannot detect the contention because the tone of ONU #1 has not reached the receiver of ONU #2 at time <c>. On the other hand, at time <d>, ONU #1 finds contention and therefore removes the tone immediately. ONU #2 holds the tone until data channel empty and then sends the data out. In this case, contention occurs and there is a winner (ONU #2).

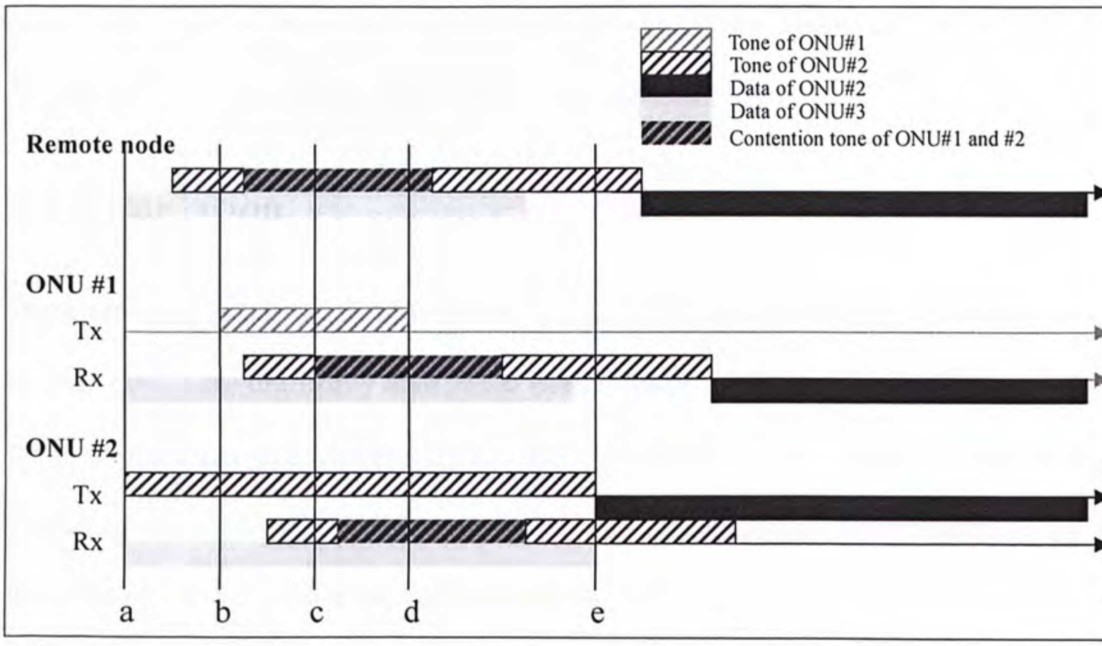

Figure 3.5 Case 3: Contention with one success

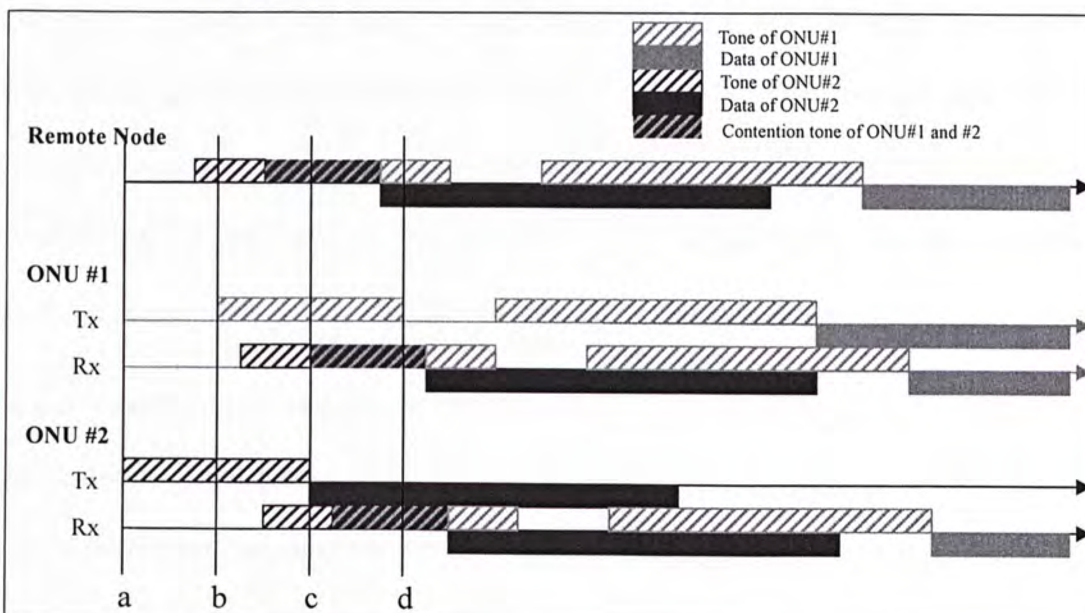

Figure 3.6 Case 4: Contention with one success

The situation is similar in the fourth case. In this example, data channel is initially empty such that ONU #2 can send data once it finds no contention at time <c>. When ONU #1 finds contention at time <d>, it removes its tone from the tone channel. In case 3 and case 4, there are contentions with one of the involved ONUs wins the contention. Data collision is not found in all the cases.

3.2. Simulation

The simulation is carried out by assuming that the distances between the remote node and the ONUs are uniformly distributed from 0.2 km to 10 km. In EPON and APON, the maximum distance between OLT and ONUs is 20km. The range we have chosen in our simulation matches the parameter in [12]. This results in the maximum round trip time of 0.1 ms. So every tone request should hold at least 0.1 ms, i.e. $T_s \geq 0.1ms$. Number of ONUs in the network being simulated is ranging from 1 to 128. Packet duration is defined as the time between the first bit and last bit of the packet being sent, which is uniformly distributed from 0.1 ms to maximum packet duration. In order to investigate the performance on using different packet size, maximum packet duration of 0.3 ms and 2.0 ms are used in the simulation. We defined packet duration factor $\alpha = \frac{T_s}{\text{Maximum packet duration}}$. Hence a smaller α value refers to larger maximum allowed packet duration. The packets entering the ONUs follow the Poisson arrival pattern. When an ONU finds contention, it waits for a random period before sending another request. Binary Exponential Backoff algorithm [19] is used to decide how the random period should be generated. Table 3.7 shows the summary of the simulation parameters and rules.

Distance from ONU to RN	0.2km to 10km
Maximum time to travel from an ONU to another ONU	0.1ms
Number of ONUs	1 to 128
Maximum Packet Duration	0.3ms, 2.0ms
Minimum Packet Duration	0.1ms (required by CSMA/CD)
Protocol Sleep Time T_s	0.1ms
Contention Resolution Scheme	Binary Exponential Backoff Algorithm Max. Retry: 16 Slot Time: 0.1ms (T_s) Max. Waiting Time for next retry: 1024 x 0.1ms
Packet Arrival Model at ONU	Poisson Arrivals
Bit Rate	Not Defined (Protocol independent)
Buffer Size	Infinite

Table 3.7 Simulation Parameter

CSMA/CD scheme which is described in previous chapter is used to compare the result with that of the proposed scheme. In CSMA/CD, each ONU keeps monitoring the data channel and stops the data transmission immediately if collision is detected. Different from the original Ethernet network, garbage code [19] will not be sent to notify other node that there is collision. This assumption is made as every ONU can detect if there is collision due to the physical architecture of PON. Minimum packet duration is needed for CSMA/CD scheme such that all collisions are detected in data sending period. BEB is used as the contention resolution scheme in the CSMA/CD being simulated. Collided packets are retransmitted after a random time which is decided by BEB. For the CSMA/CA case, since collision is avoided, there is no minimum packet duration requirement. The simulation is implemented using Java.

3.2.1. Effect of Loading on Network Utilization

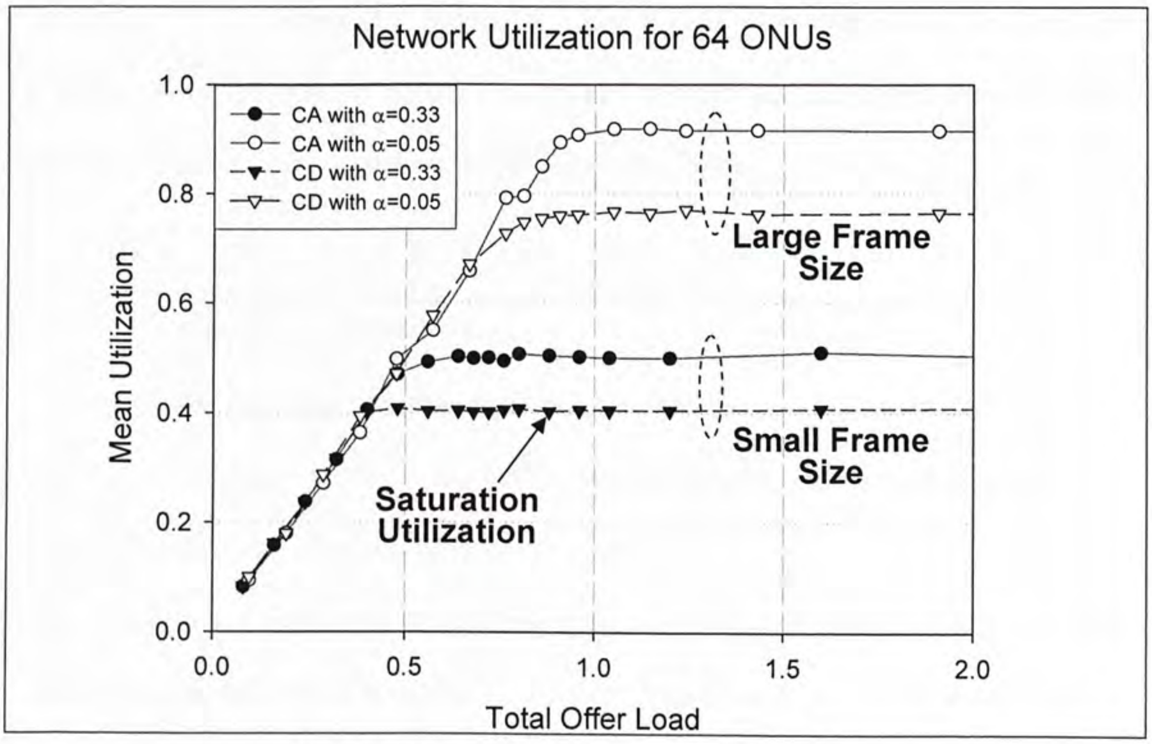

Figure 3.8 Effect of Loading on Utilization

By keeping network size to 64, the effects of offer loading on utilization are studied and plotted in Figure 3.8. The utilization reaches plateau at high loading. In other words, the network has an upper limit on utilization, and we call this limit saturation utilization. When offer load approaching the saturation utilization, data begins to accumulate at input buffer of the ONU, resulting in buffer overflow and eventually data loss.

When comparing CA and CD schemes, CA scheme has higher utilization under the same network size and α value at high loading (higher saturation utilization). A small α value shows better performance in terms of utilization. Different from CD, the decision of contention winner in CA can be done during an ONU is sending data for the CA scheme. This account for less wasted time slot and therefore higher utilization can be achieved. Saturation utilization can be estimated if we know the distribution of the time to decide a contention winner and the distribution of packet duration (depends on α value) by the following formula.

$$\boxed{U = \frac{P}{\max\{P, C\} + D}}$$, where

U = Saturation Utilization

P = Average Packet Duration under heavy loading

C = Average time to decide next contention winner under heavy loading

D = Average traveling time from ONU to RN

The denominator represents the average time between two consecutive packets that are sent under saturation condition. Average packet duration P is a function of α. With a larger α value, C is larger under heavy loading and hence the saturation utilization is smaller. In practice, C depends on many factors, such as the distance of each ONU from RN, network size, network loading, etc.

3.2.2. Effect of Network Size on Utilization

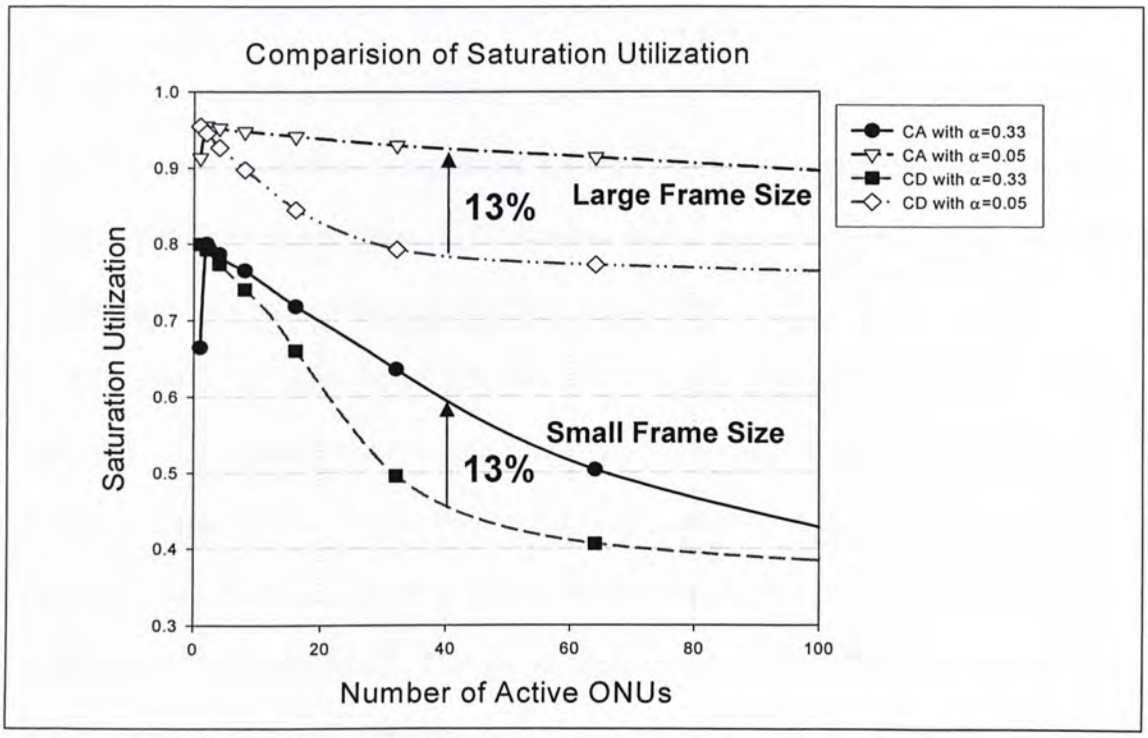

Figure 3.9 Effect of Network size on Saturation Utilization

There are more contentions should there be more competitors on the networks. The increase in C will eventually lead to lower utilization, as shown in Figure 3.9. Saturation utilization as seen in above section provides a good way to measure the effect of network size. In the simulation, the information of saturation condition is collected by setting all ONU busy.

For a special case when there is only one ONU on the network, the saturation utilization of the protocol can be estimated by $\left(\frac{P}{P+T_s}\right)\times 100\%$. Before sending every packet, that ONU sends a tone request and then waits for T_s. When there are two ONUs on the network, assuming these two ONUs are busy, one can always win the next time slot when the other one is sending data. Hence the protocol can reach its theoretical maximum, and can be estimated by $\left(\frac{P}{P+D}\right)\times 100\%$.

In both CA and CD schemes, higher number of active ONUs leads to a decrease of utilization due to keener competition among ONUs. Except from the case when there is one ONU on the network, CA scheme shows higher saturation utilization than CD scheme does on network size between 2 and 100.

In 3.2.1 and 3.2.2, we have shown and discussed that CA has more effective use of time slot. CD scheme handles collision of data by resending the collided packet after a random backoff time. Hence a proportion of time slots is wasted for collided data packets. In CA scheme, there is still contention for channel resource, but it provides collision-free characteristic. The decision for owner of next time slot can be done when some ONUs are sending data. On the other hands, since the proposed algorithm is not contention-free, when comparing with MPCP of EPON, it still shows a smaller utilization. This is explained by mean length of contention period C is larger than mean packet duration P. In MPCP, C is a constant and P is usually dynamically adjusted, such that nearly 100% utilization is possible by setting P larger than C.

3.2.3. Delay Performance

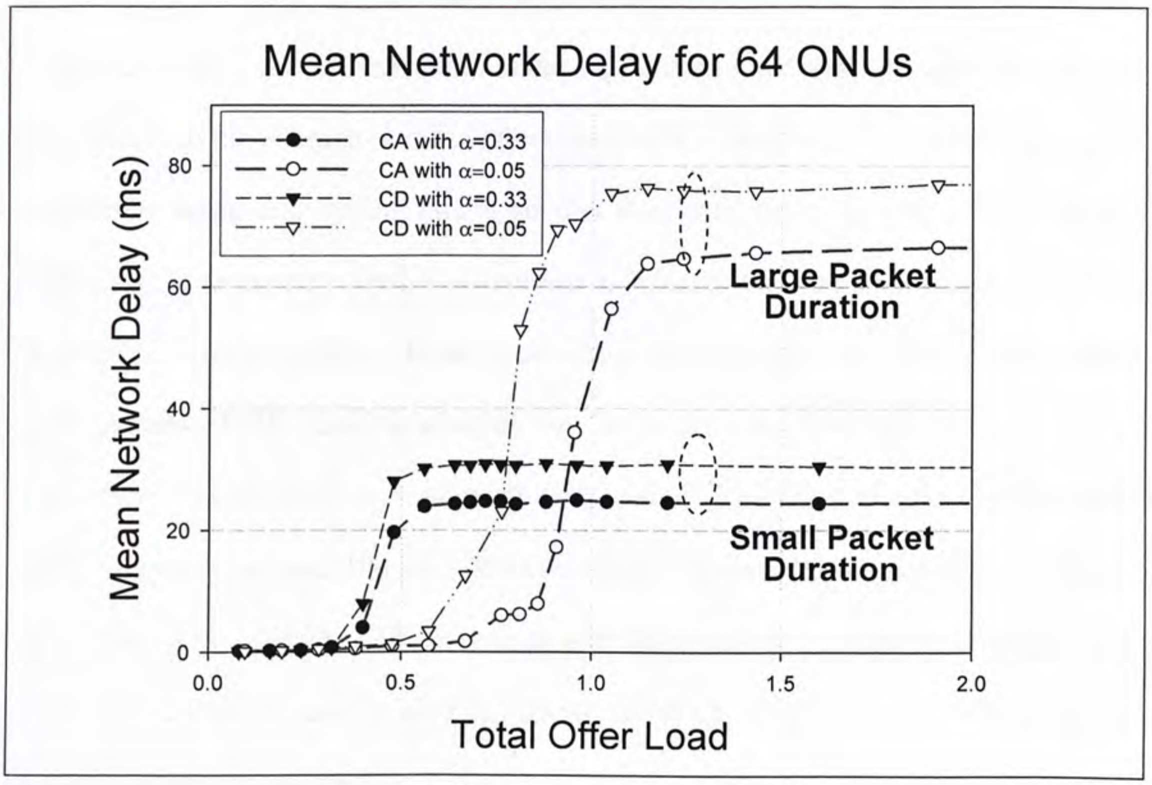

Figure 3.10 Effect of loading on delay

The delay-loading graph (Figure 3.10) shows the effect of loading on network delay, which defines as the duration between a packet entering the head of buffer and the first bit being sent. Another definition of delay, the overall delay, is predictable from the network delay distribution and input rate by modeling the buffer into an M/G/1 queue, as we assume that the incoming traffic follows Poisson arrival pattern. Average overall packet serving time (overall delay + packet sending time) can be obtained from the mean overall delay and mean packet sending time (uniformly distributed in our model).

The delay-loading curves show nearly zero delay (~100 μs) when offer loading is small. The result is reasonable as contention rarely happens when loading is small. Under high loading, the network delay curves become flat. This is because all ONUs are active (not in idle state) under high loading and average network delay is dependent on the average number of active ONUs. However, the overall delay will be infinite when data outing rate is smaller than data incoming rate, where infinite buffer size is assumed. Buffer is assumed to have infinite size in our simulation, but it is not possible in reality. Data is lost when buffer overflows. The system is said to be unstable if offer loading is higher than or close to saturation utilization.

CD scheme shows nearly zero network delay under light load, while CA scheme using pilot tone shows around 100 μs. When loading is higher, CD scheme shows a higher delay than CA scheme. Therefore, an adaptive scheme is possible to enable CD scheme under light loading and CA scheme under high loading, but it will make the protocol more complicated.

Figure 3.11 shows the network delay distribution under different loading and α value for a network size of 32 ONUs in one simulation trial. The 100 us peak in (a), (b) and (c) represent the proportion of packets which encounters no contention and empty data channel. In (a) and (b), where network loading is small (~0.3), the probability of a head of buffer packet can be sent in 100 μs is over 50%. Under high loading (~0.9), this probability is smaller than 0.10 as shown in (c) and (d). If network is busy, it is normal that a new arrival has to wait longer until it can be sent; also, the data channel is occupied most of the time. The relatively dense distribution in 101 μs to 2100 μs of case (a) and (c), and 101 μs to 400 μs of case (b) and (d), are explained by the time to wait for an occupied channel to become available.

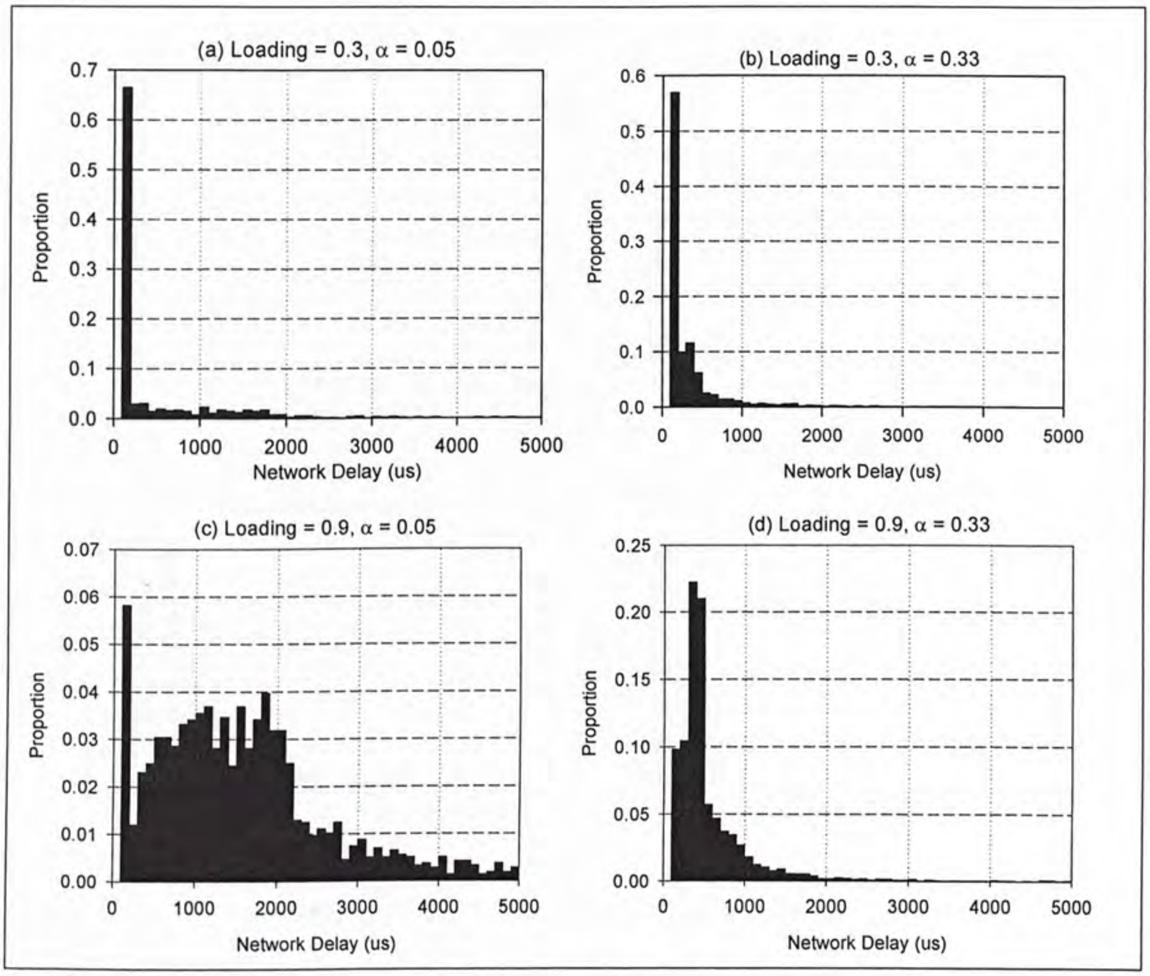

Figure 3.11 Network Delay Distribution

We further investigate the protocol activity under different loading. Figure 3.12 shows the number of retry of each successfully sent packet experienced. In general, under light loading (<0.3), the probability of success in first attempt is over 0.8, as shown in (a) and (b). Contention is less likely to happen under a small network loading. This results in small network delay. Under high loading (~ maximum utilization), average number of retry recorded is 2.36 in case (c) and 1.28 in case (d). Probability of network loss (retry >16) is small in all the cases. Packet loss is mainly due to buffer overflow as the incoming rate is higher than the departure rate.

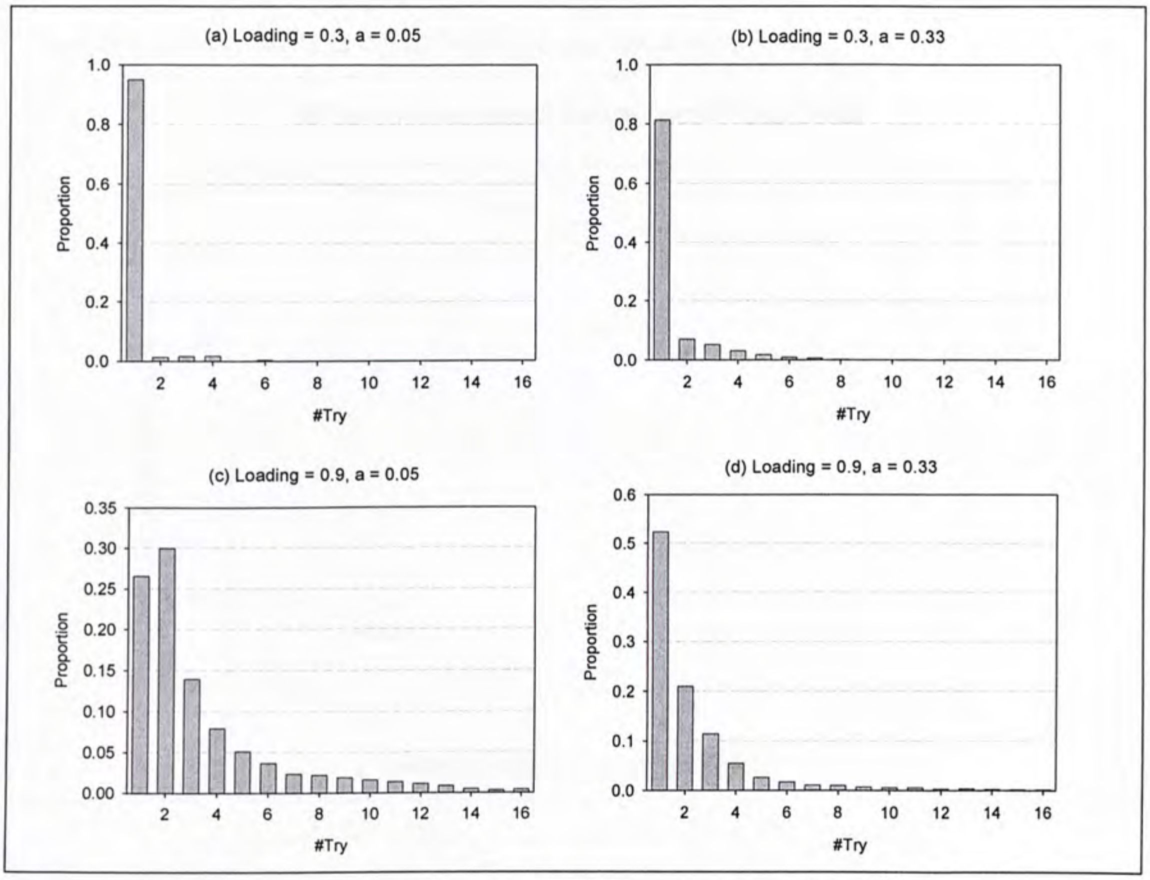

Figure 3.12 Distribution of retry number

3.2.4. Effect of Distance from Remote Node

In previous simulation, we have assumed that the distance between the ONU and optical power splitter is uniformly distributed between 0.2 km and 10 km. As specified by the protocol, an ONU will first check the tone channel before it sends the request tone. Hence, if the average travel distance among the ONUs is shorter, there is a higher chance to detect non-empty tone channel under high loading. Occurrence of contention (Case 2) can be reduced and thus it is expected that the saturation utilization is higher in this case. It is interesting to see how the average distance from RN to ONU affects the network performance with fixed T_s.

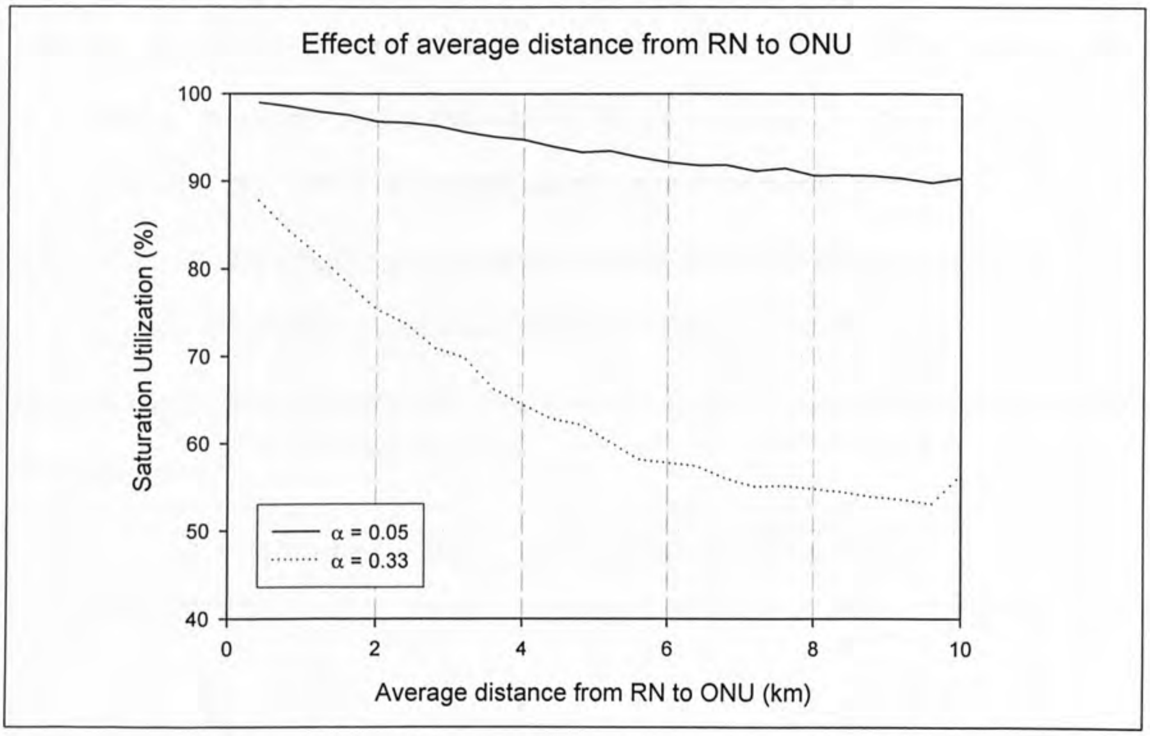

Figure 3.13 Effect of average distance from RN to ONU

Figure 3.13 shows the simulation result of protocol performance against average travel distance. Network size is 32. The simulation result shows that the saturation utilization decreases as ONUs are further away from RN. A larger α value also shows a faster decreasing. Hence we should make the RN-ONU distance as small as possible when this protocol is used on PON.

3.2.5. Effect of Maximum Packet Duration on Utilization and Delay

As we have seen in above, the α value plays an important role in deciding the protocol performance. Simulations have shown that a smaller α value gives a better performance. In our simulation, the α value alone decides the distribution of frame time. The proposed protocol does not need to specify the minimum duration of a packet (in simulation, as we want to compare the performance against CD scheme, minimum packet duration of 100 μs is assumed). Since the choice of α value will not affect the functionality of the protocol, a small α value is preferred as it gives higher utilization and smaller delay, but we also need to consider the network delay during saturation scenario. Figure 3.14 shows the effect of α value on saturation utilization, showing a small α gives higher network delay under heavy loading

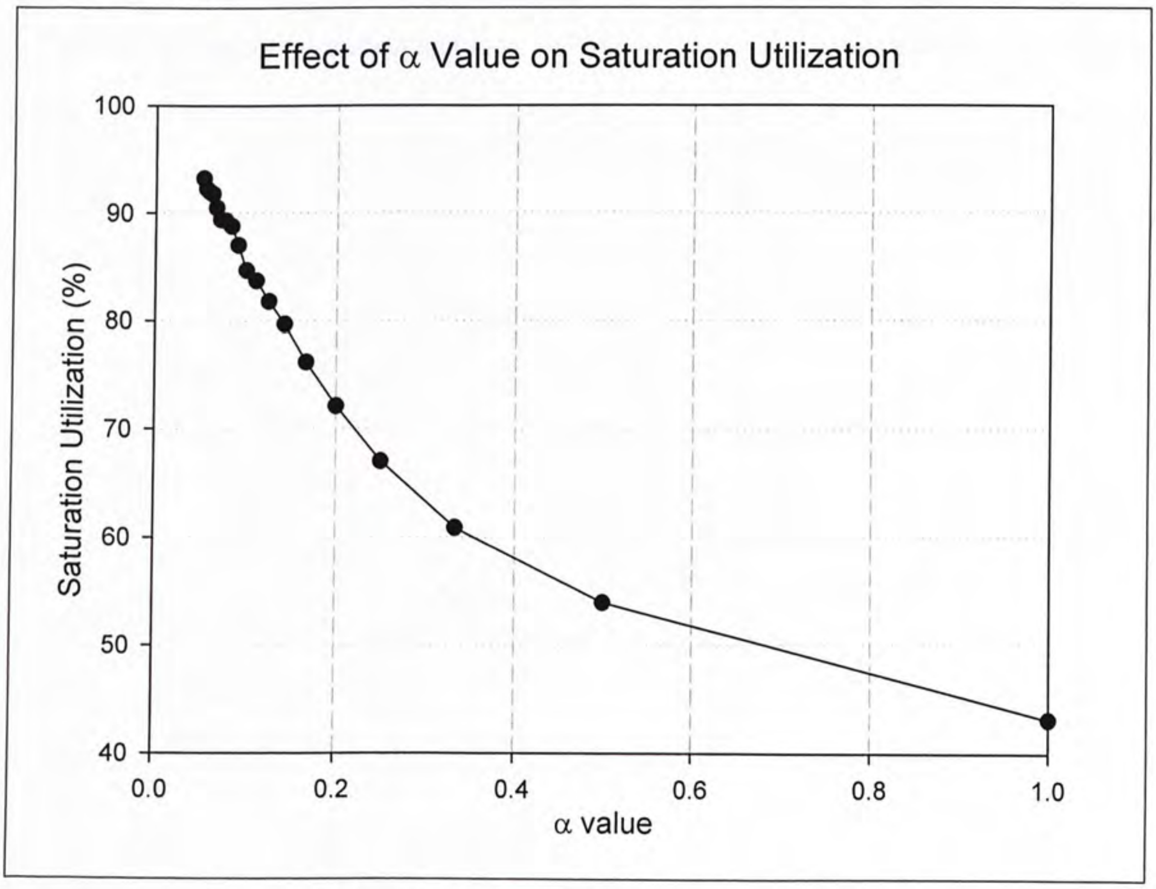

Figure 3.14 Effect of α value on saturation utilization

3.3. Conclusions

In this chapter, we present a remedy based on carrier-sensed multiple access mechanism (CSMA) with collision avoidance (CA) using Pilot Tone to resolve the upstream contention problem in a TDMA power-splitting passive optical network. Synchronization among ONUs and OLT is not needed. Simulations have been implemented using JAVA SDK and the results show that the proposed scheme can give higher utilization and smaller delay when comparing with CSMA/CD scheme. The effect of packet duration and mean RN-ONU distance are discussed.

Chapter 4:

Protocol Enhancement on Various Aspects

In the previous chapter, we have proposed to use pilot tones, which are permeable to optical data, to enable optical signaling (carrier-sensing) and achieve collision avoidance (CA), instead of CD. We have shown that channel utilization over 90% is possible when there are 32 ONUs in the network, comparing with only 70% for the conventional CSMA/CD scheme. In this chapter, various protocol enhancements are implemented to achieve different goals: higher utilization, eliminating capture effect brought by BEB, and class of service (CoS) support.

4.1. Utilization Enhancement

In the previous chapter, we have shown that a smaller α value (i.e. larger packet per request) can achieve higher utilization and smaller delay. If there are less requests, the time to have a contention winner, C, is shorten due to less competitors in one contention process. Hence, higher saturation utilization is obtained as discussed in section 3.2.1. One variation of the protocol is to add a decision stage in the previous algorithm. In the modified protocol, ONU can send more than one packet with one request when the channel is available. This modification can be done by sensing the tone channel after sending a packet. When an ONU finishes sending a packet, it checks whether the tone channel is empty. If tone channel is empty, it implies that there is no other ONUs waiting to send data after it has finished sending the current packet. Therefore, the ONU can continue to send another packet if it has another packet waiting in its buffer. The modified algorithm is shown in Figure 4.1.

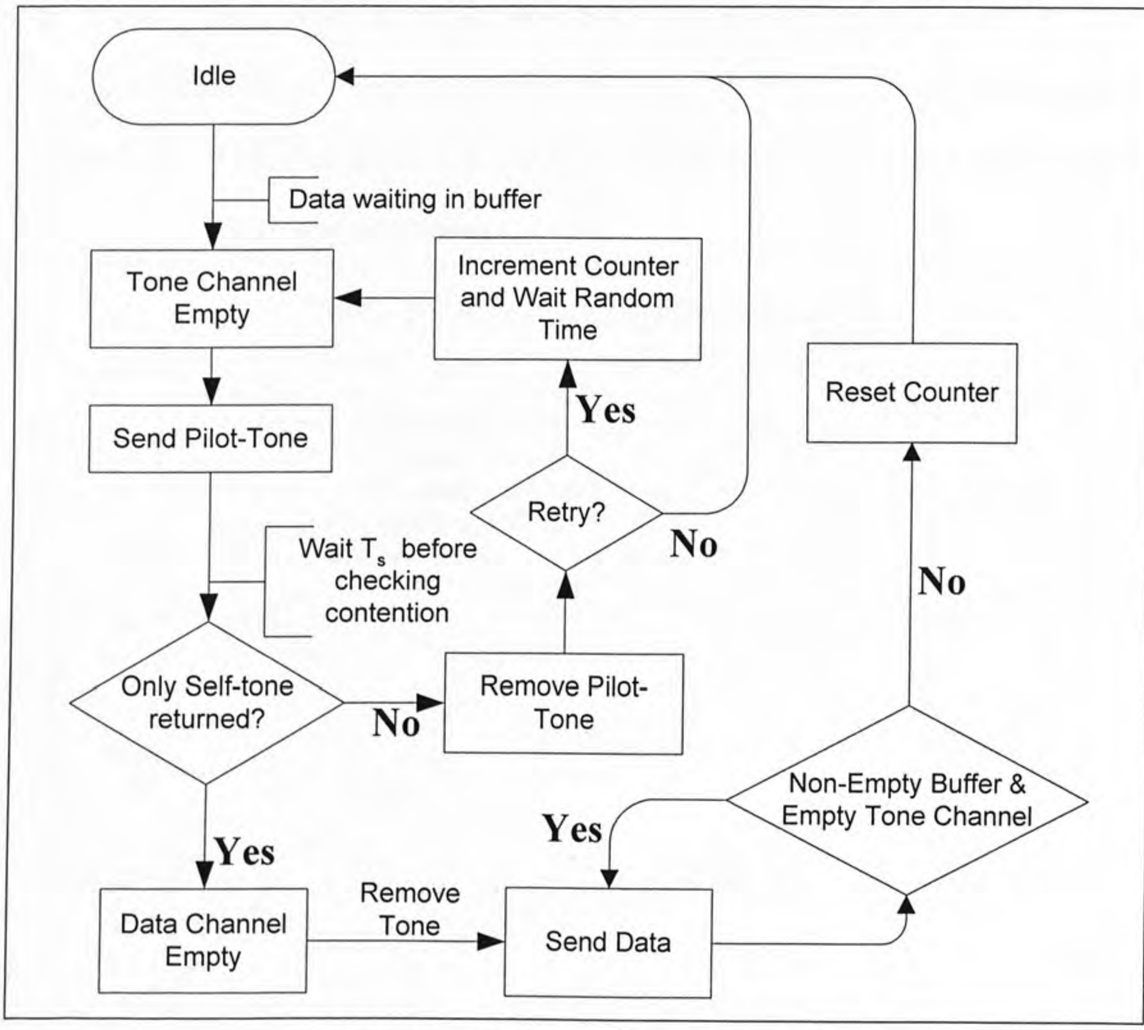

Fig. 4.1 PON CSMA/CA Algorithm

The added decision stage reduces the request rate. The channel utilization is therefore increased by allowing an active ONU to utilize an unused time slot. However, the adding of such stage may lead to higher complexity of the protocol. We will see how this modification affects the utilization and delay if the complexity is not an issue.

4.1.1. Improvement on Network Utilization

As the modified protocol reduces the rate of making request, it is expected that the saturation utilization will be higher than that of the original scheme, under the same protocol parameters. Figure 4.2 shows the simulation result on utilization against loading for PON with 32 ONUs. We can see from the figure that the modified protocol provides higher utilization.

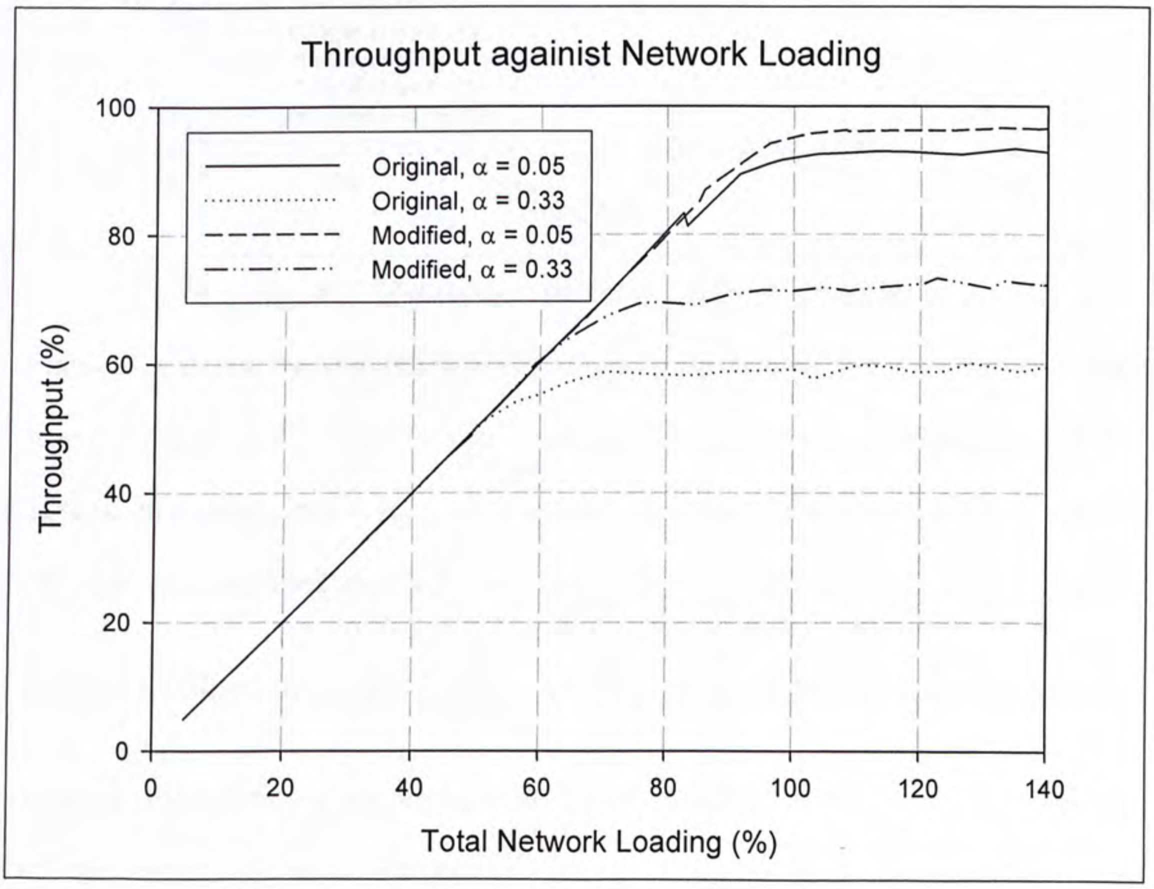

Figure 4.2 Utilization against network loading

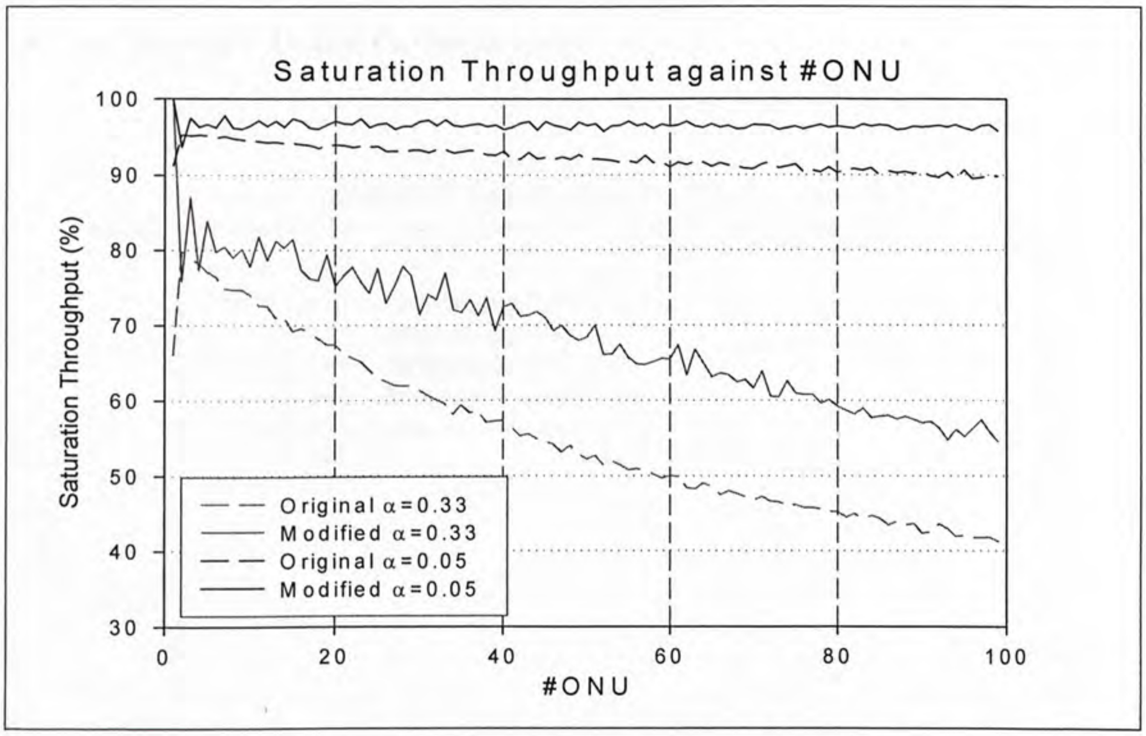

Fig. 4.3 Saturation utilization against Network Size

Figure 4.3 shows the simulation result of saturation utilization against network size when α=0.33 and α=0.05. The improvement on maximum utilization is about 12% when network size is 40. For a special case when there is only one ONU on the network, the modified protocol can reach 100% utilization, whereas the original protocol's utilization is only $\left(\frac{P}{P+T_s}\right)\times 100\%$. When there are two ONUs on the network, the utilization performances of the two versions are the same. For the rest of the cases, the modified protocol shows a satisfactory improvement. The improvement for α=0.33 is greater than that of α=0.05. This modification enables the protocol to use a large α and still exhibits large utilization.

4.1.2. Network Delay Performance

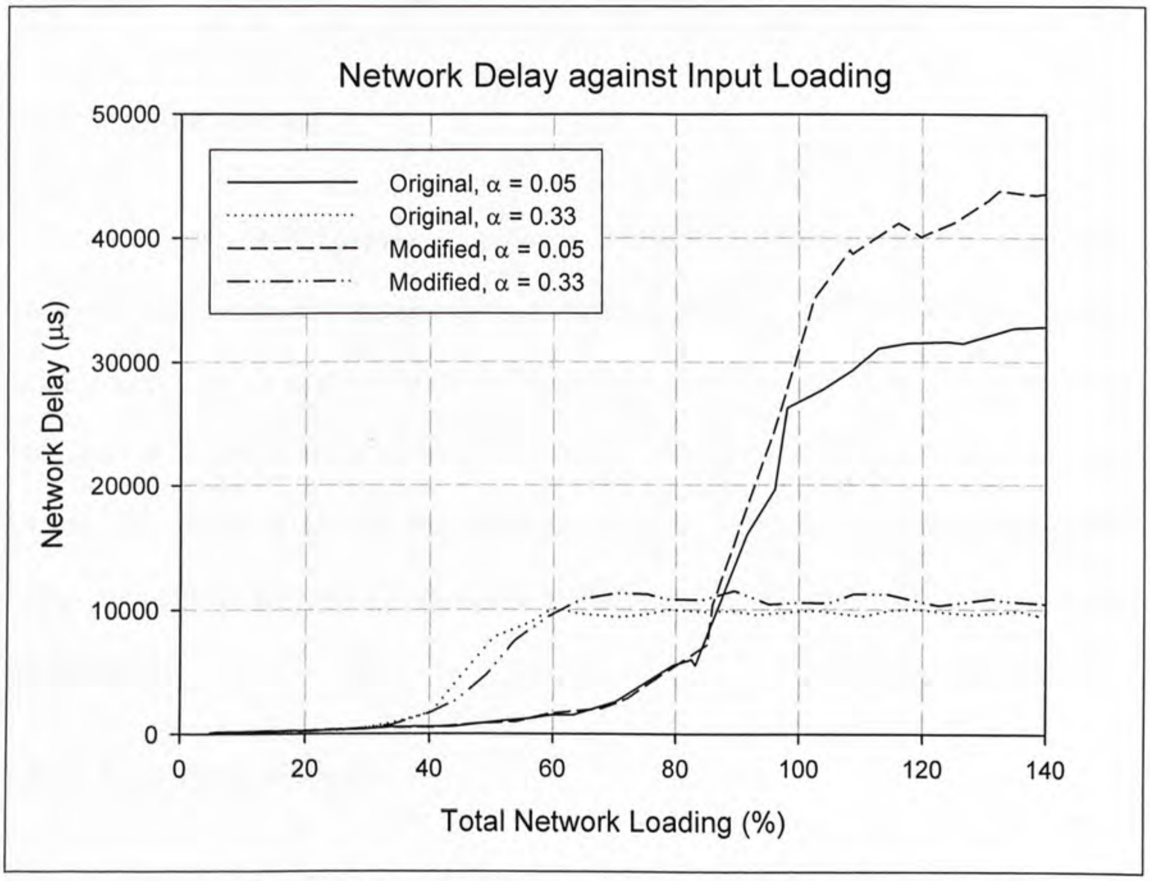

Figure 4.4 Network Delay against input loading

Figure 4.4 shows the network delay on different loading for a network with 32 ONUs. Under light loading, the performance of the original scheme and the modified scheme are about the same. This is because the probability of having two or more packets in the buffer is small when loading is small. Hence, the added state is skipped most of the time. The modified protocol therefore works almost in the same way as that of the original protocol. Under high loading before saturation, the modified protocol shows a smaller network delay on average. This is because there are packets sent without making request. Statistically speaking, the average network delay is reduced.

When the network is saturated, the modified protocol has a larger delay. This shows that packets have to wait longer on average before it can win a contention.

4.1.3. Conclusions

The modified protocol provides a possible means to increase the channel utilization. Marked improvement is made when α value is large. In some cases, large packet size (small α) is not preferable. Hence we may use the modified protocol to increase the channel utilization in these cases. However, if we are allowed to use a small α value, it is not advisable to use the modified protocol because the improvement is not significant and the modification will make the protocol more complex.

4.2. Capture Effect

The proposed algorithm uses Binary Exponential Backoff (BEB) algorithm as the contention resolution scheme. When an ONU finds contention, it waits for a random period of time before making another request. BEB algorithm is used to decide how this random period should be generated. One disadvantage of using BEB algorithm is the “capture effect” which occurs when the network loading is high. The problem originates from the handling of the retry counters. Each ONU updates its retry counter independently after a transmission attempt. Only the winner resets its retry counter to zero. This approach benefits a single busy ONU, permitting it to "capture" the network for an extended period of time.

Capture effect can also be found in Ethernet network where Truncated BEB [19] is used as the contention resolution scheme. By using different contention resolution scheme such as Uniform Backoff, the capture effect can be eliminated. Binary logarithmic arbitration method (BLAM) [20] is proposed in 1994 to solve the capture effect by having some network units (but not all) using different resolution scheme. It has backward compatibility to BEB in existing network and is added as an optional feature to Ethernet standard by IEEE802.3w working group.

4.2.1. Solution by Varying T_s

In this thesis, a solution is proposed to solve this problem on the proposed protocol. Simulation results show that an ONU will have higher chance to win a contention if its sleep time (T_s) is longer. The key idea of the proposed solution is to give those ONUs with larger retry counter a higher probability of winning contention. This is done by making T_s as an increasing function of retry counter. In order to evaluate the feasibility, we set T_s equal to (100 μs × min(#retry, 5)). Figure 4.5 shows the simulation result with 8 ONUs on the network and $\alpha=0.05$, assuming all ONUs are always active and are equidistance from the remote node (5 km).

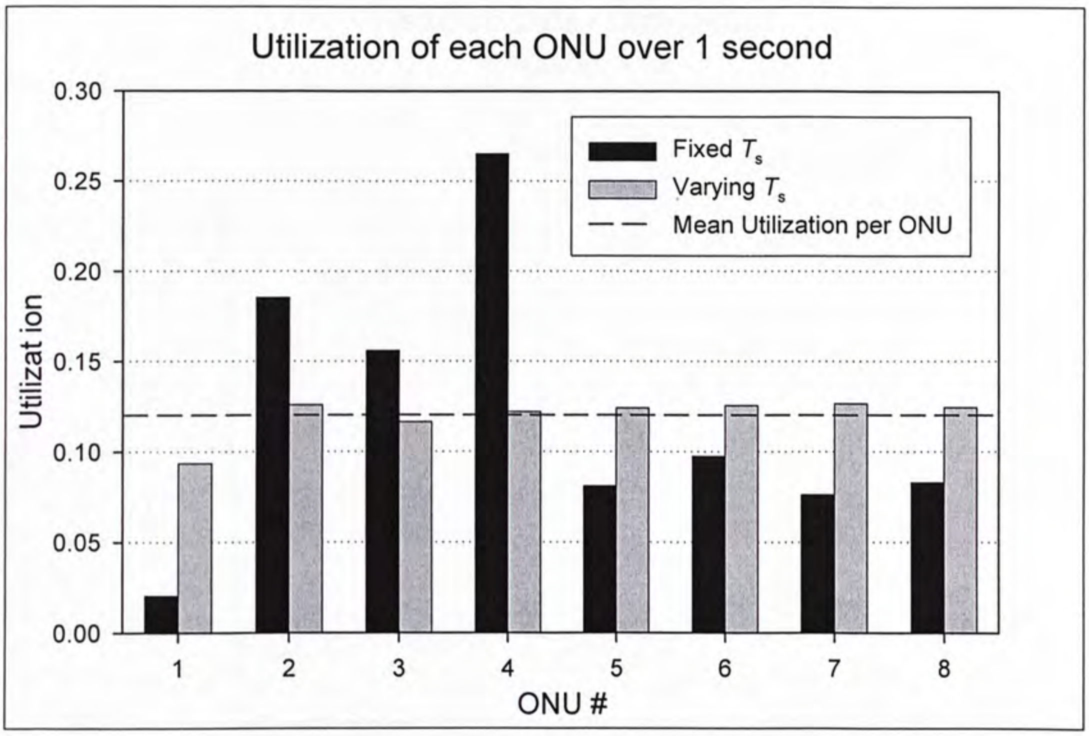

Fig. 4.5 Simulation on Utilization of 8 ONUs over 1 second

4.2.2. Simulations

In the simulation, capture effect is observed from large variance of utilization of each ONU over a short period. In the trial of figure 4.5, we can tell that ONU #1 is the victim of capture effect in this second by its low utilization and large delay experienced. ONU #4, #2 and #3 seem to be the winners. By having T_s variation, the capture effect becomes less observable, as depicted by the grey bars in figure 4.5

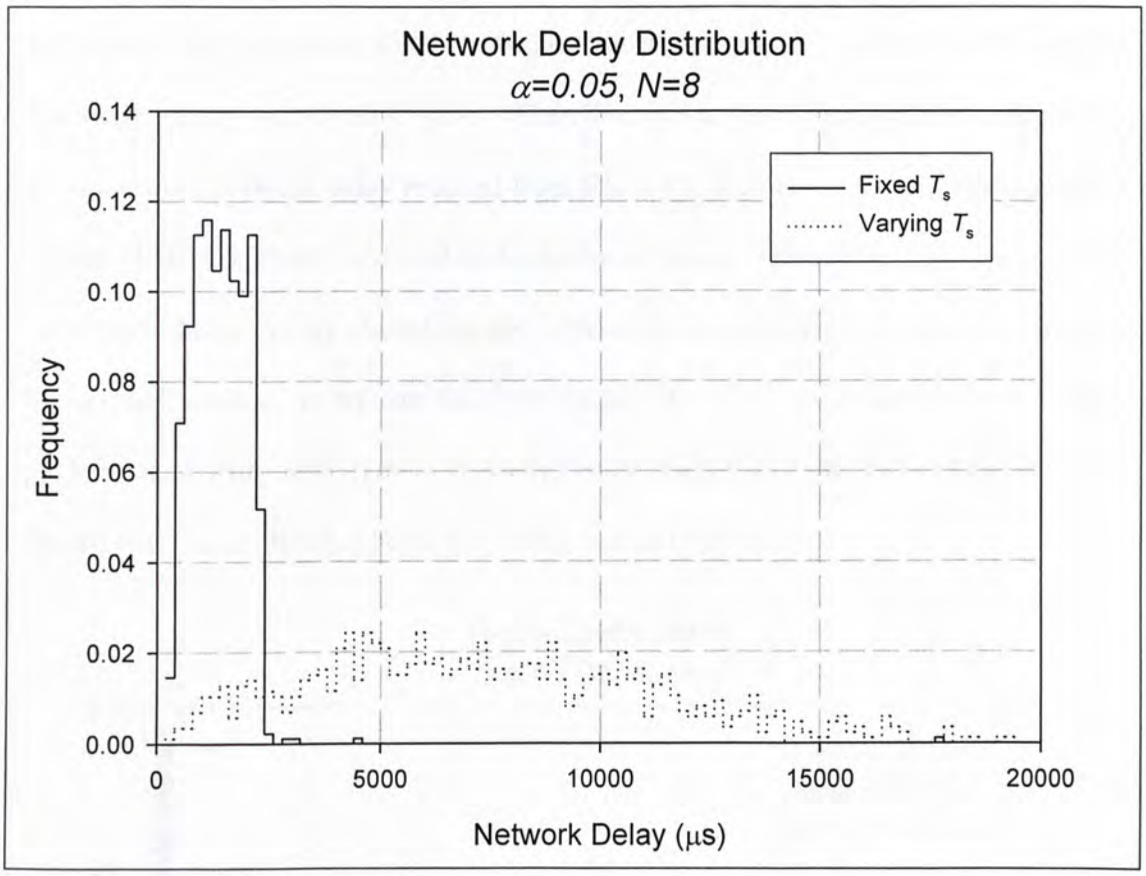

Figure 4.6 Network Delay Distribution during Saturation over 1 second

We can also investigate the capture effect by studying the distribution of network delay under saturation. Figure 4.6 and figure 4.7 show the network delay distribution and retry number distribution, respectively. With fixed T_s, the average network delay is smaller. Most of the packet is sent with zero retry. On a busy network, every network unit is trying to send at the same time. It seems not reasonable that most of the packets are sent without contention (zero retry). From the simulation traces, it is found that most of the packets that are successfully sent over the time period are from a small group of ONUs. So we conclude that there is capture effect under high loading.

By varying the T_s, the distribution of network delay and retry become flatter. ONUs with higher retry counters now have higher priority to send. Hence network capacity can share evenly among all ONUs. The effect of varying T_s can be also reflected by the maximum network delay reduced from 626 ms to 8.25 ms in the proposed solution. Capture Effect is greatly reduced in the revised scheme. However, since the network capacity is more evenly shared among ONUs, the average network delay is larger in the revised scheme, as we can see from figure 4.6. This is because there are fewer packets sent with zero retry. However, with larger average network delay, it is shown that the modified scheme has lower channel utilization.

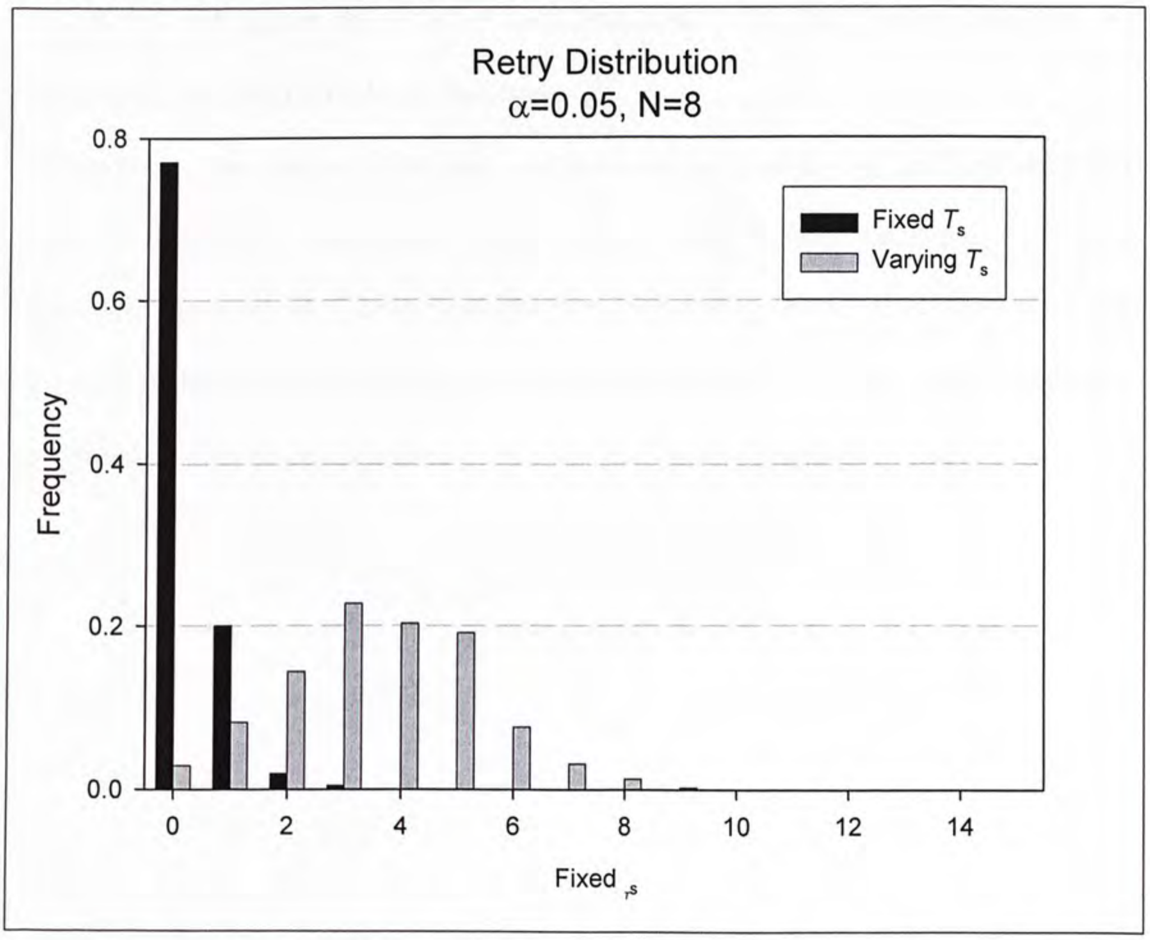

Figure 4.7 Distribution of Number of Retry Before Winning Contention

4.2.3. Conclusions

The capture effect induces fairness under heavy loading. In this section, a possible solution by varying T_s is proposed, which is different from the conventional contention resolution scheme such as BLAM and Uniform Backoff. The difference of using this solution from the conventional solutions is that it gives higher priority to packets which experience more failures in request. Hence packet loss due to too many failures can be reduced. The trade-off of this solution is that it gives lower utilization and higher delay under saturation case. We may further fine-tune the sleep time function to moderate this trade-off.

Nevertheless, the capture effect and saturation can be prevented by traffic shaping and traffic monitoring in the upper layer such as leaky bucket technique. By such schemes, the network loading can be kept under the saturation utilization, at the expense of higher complexity in the upper layer protocol. In this section, we have provided an alternative to solve the problem in a "protocol-specific" way.

4.3. Introducing CoS to existing network

We can characterize the performance of a packet-based network by several parameters: bandwidth, latency (packet delay) and jitter (delay variance). A network providing quality of service (QoS) is able to provide bounds on some or all these parameters on a per-connection basis. Some networks cannot give these guarantees, for example Ethernet is handicapped by the random nature of binary exponential backoff algorithm. Hence in these networks, QoS can hardly be implemented. In order to support diverse application requirements, such networks separate all the traffics into a limited number of classes and provide differentiated service for each class. Such networks are said to maintain classes of service (CoS).

To support CoS, the network must be able to classify traffic into CoS and provide differentiated treatment to each class. IEEE P802.1p "Supplement to MAC bridges: traffic class expediting and dynamic multicast filtering" maps traffics classes into different priority queues [21]. This standard extension separates the network traffics into seven classes: network control, voice, video, controlled load, excellent effort, best effort, and background. It also defines how we should group the traffic when less than seven classes are allowed. In our proposed scheme, we assume there are three classes.

4.3.1. Principle

The support of CoS is through two parts: intra-ONU scheduling and inter-ONU scheduling. Intra-ONU scheduling divides incoming packet into priority queues.

Packets in the highest priority queue are sent with highest priority within the ONU. This ensures packets with higher priority classes will have a lower delay and better utilization than lower priority classes. Inter-ONU scheduling allocates the time slot to different ONUs according to the packet type that ONUs is sending. It is expected that more important packets should have higher priority in the scheduling. In IEEE802.3ah, where MAC message is used, inter-ONU scheduling is done in the OLT which allocates time slots for each ONU. Since the OLT has the knowledge of buffer status of all ONUs, it is easier to allocate time slots with appropriate priority given. Therefore, CoS can be easily implemented in OLT by time slot allocation algorithm.

On a distributed system, inter-ONU scheduling is done by all ONUs. One difficulty is that no ONU has the knowledge of buffer status of other ONUs. Hence the request signal should include the priority information.

The key idea of CoS provisioning in proposed protocol is similar to the solution proposed for capture effect in previous section. Packet with higher priority will be given a longer "sleep time", T_s. Hence, packets in higher priority queue have a higher chance to win contention. The combined effect of intra-ONU scheduling and inter-ONU scheduling is simulated with three service classes.

4.3.2. Simulation Model

In our simulation model, intra-ONU scheduling places packets of different classes in different queues. Before an ONU involves the contention process, the algorithm will check the status of the queues to decide the length of sleep time, T_s. As mentioned in previous section, if the algorithm is going to send a packet with higher priority, longer T_s will be used. The simulation model defines T_s as follows:

Packet with lowest priority (Class 1): $T_s = 100$ μs;

Packet with medium priority (Class 2): $T_s = 200$ μs;

Packet with highest priority (Class 3): $T_s = \min\{800, 200 + retry \times 100\}$ μs.

Hence, class 3 packet has the highest priority to send. Since capture effect is likely to happen when network loading is high, we adopt the capture effect solution of section 4.2 when the algorithm is trying to send a class 3 packet. By this assignment, it is expected that sending a packet with higher priority will have a higher chance to win contention.

We define the loading and usage of each class: Class 3 packets are used for circuit-over-packet based connection to support services such as VoIP. This type of service usually has constant bit rate. Class 2 packets emulated services can support Variable Bit Rate (VBR) such as MPEG-coded video. Class 1 packets have the lowest priority to send. This priority is used for supporting non-real-time data transfer such as FTP, HTTP services.

When we vary the total loading in our simulation, we always keep the Class 3 services utilizing 10% of the total channel capacity. The remaining load is shared equally by Class 2 and Class 1 services. This distribution is used in order to compare the result obtained from IPACT in [21].

4.3.3. Utilization Performance

Figure 4.8 and 4.9 show the utilization status of channel under different loading.

Under light loading (< 40%), utilization is the same as input loading. Packet loss is not likely to happen. Under high loading, when the protocol cannot support all the input packets, Class 1 packets are displaced by higher priority classes. We can see in the figures that the utilization of class 1 packets is smaller than its input rate under heavy loading. Before the utilization of class 2 packets reaches the maximum, Class 3 and Class 2 packets can still remain in-out equilibrium. In figure 4.9, where α =0.33, we observe that there is a peak around 45% loading. Beyond this loading, the total utilization drops. Under high loading, most of the ONUs are trying to send Class 2 or Class 3 packets. This lengthens the time for the contention process and results in lower utilization of the channel.

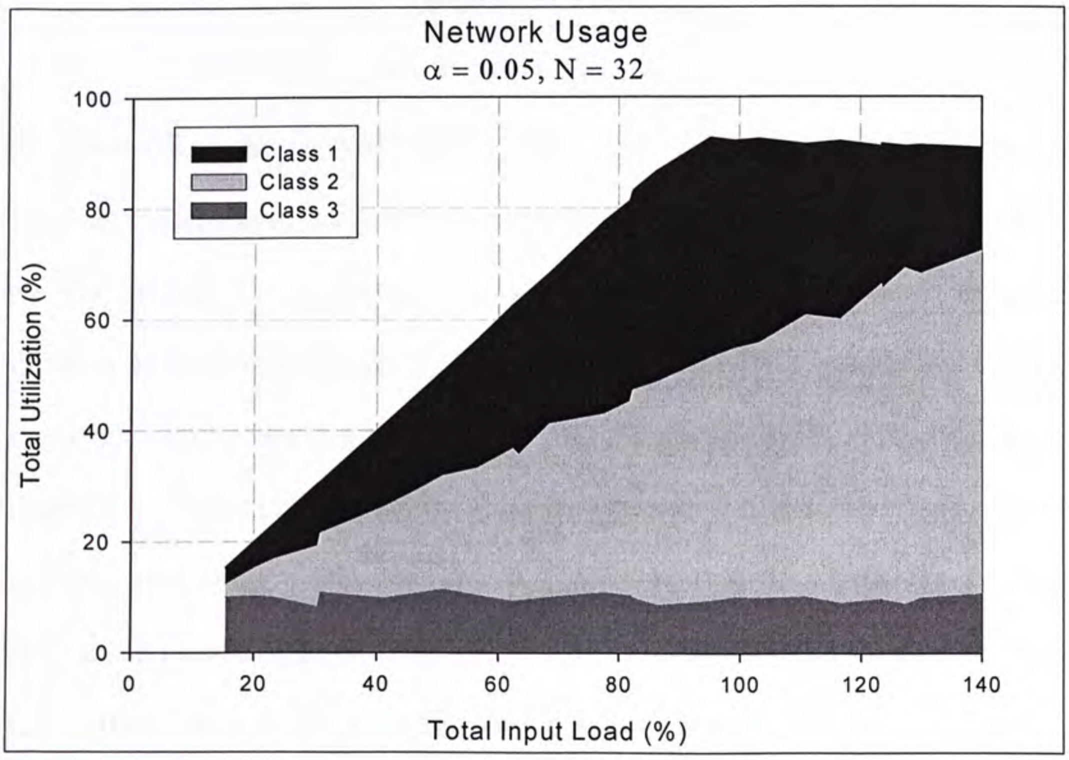

Figure 4.8 Network Usage for α =0.05

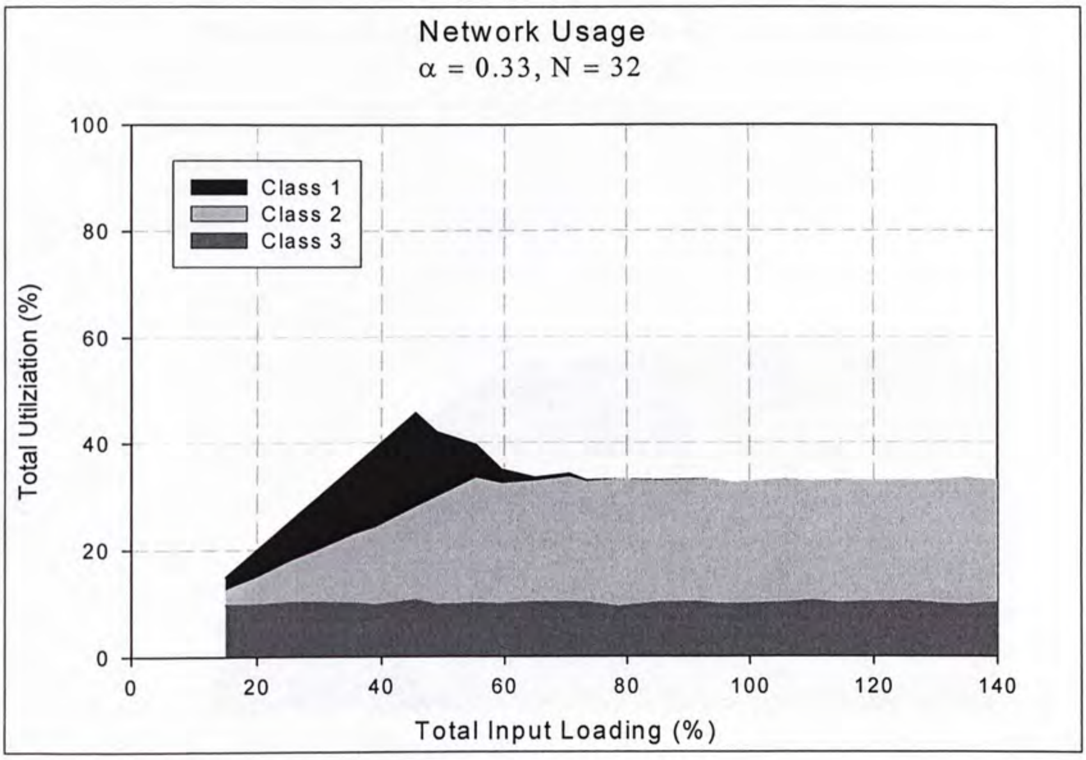

Figure 4.9 Network Usage for α *=0.33*

The above effect can be attributed to both inter-ONU and intra-ONU scheduling. Since many works have been done on intra-ONU scheduling, we are interested to see how the inter-ONU scheduling affects the utilization. Figure 4.10 shows the utilization of each class for α=0.33 without using inter-ONU scheduling, i.e. T_s of sending all classes are set to 100 μs. The simulation result shows that higher utilization is achievable because the time for contention C is smaller than that with inter-ONU scheduling. However, the simulation shows great improvement for class 3 and class 2 packets in term of delay which will be discussed in next section. For a small α (0.05), the utilization performance with and without inter-ONU scheduling is about the same.

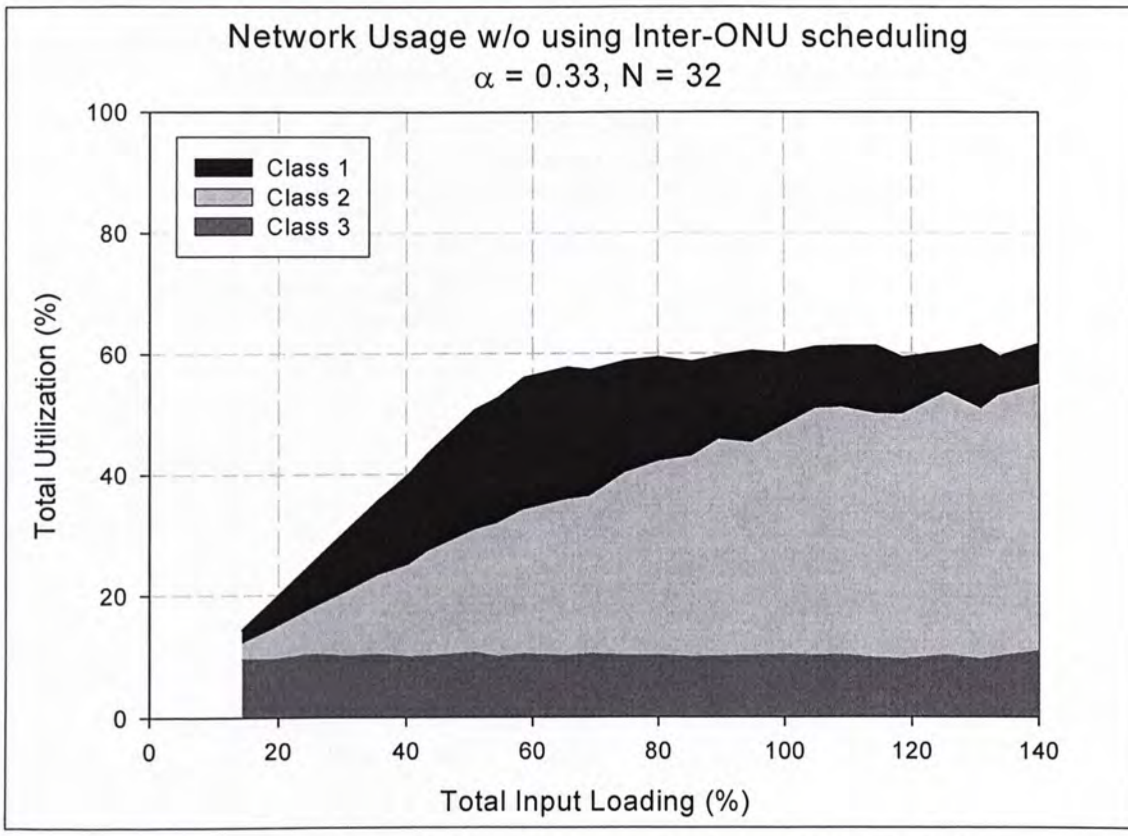

Figure 4.10 Network Usage without using Inter-ONU Scheduling

4.3.4. Delay Performance

Figure 4.11 shows the total delay (sum of delays in buffer and network) when both intra-ONU and inter-ONU scheduling are used with α=0.05 and N=32. Class 3 packets can always be sent with average delay less than 2 ms, regardless of the network loading. At light loading, the delays experienced by three types of packets are about the same. Since Class 2 packet has higher priority than Class 1 packet, under heavy loading, there is more proportion of Class 2 packet being sent. The input rate of Class 1 service is higher than its utilization under heavy loading. Hence services that use class 1 packet become unstable. This is shown by the large delay of class 1 packet under heavy load.

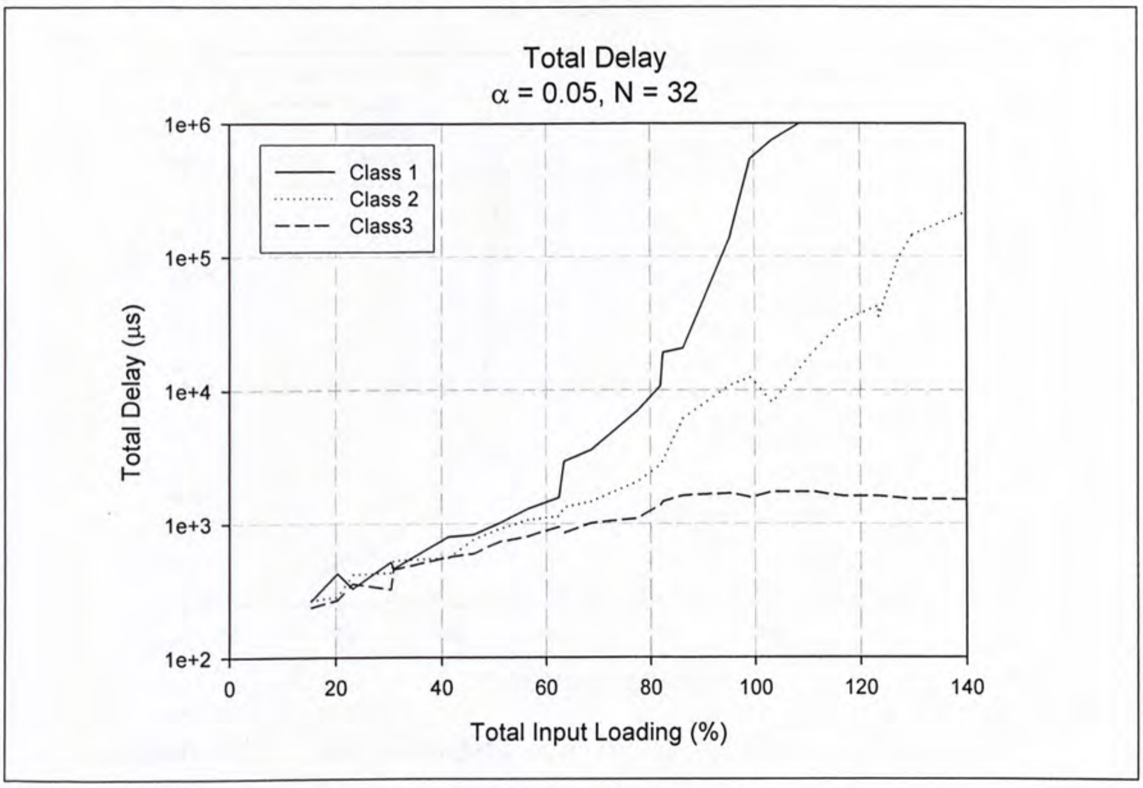

Figure 4.11 Total Delay for 3 different classes of service

For a large α, the result is similar, except that saturation will occur at a smaller loading, as shown in figure 4.12. When network is saturated, Class 3 packets still occupy 10% of total channel utilization. All the remaining output of the network is Class 2 packet. Due to this reason, the figure cannot shows the delay of class 1 service under high loading as there is no sufficient data (no class 1 packet can be sent during the simulation period). The simulation results show that the low delay of class 3 service is not affected by α value.

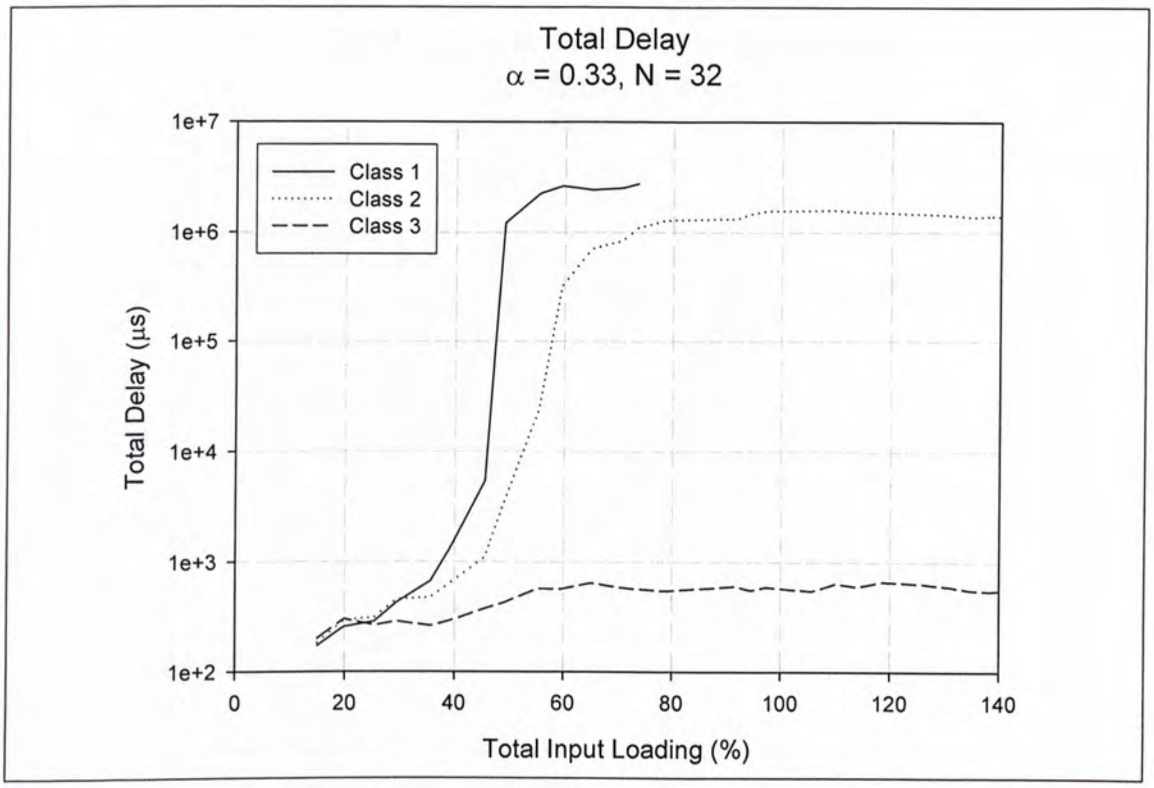

Figure 4.12 Total Delay for 3 different classes of service for α =0.33

The simulation result shows the combined effect of intra-ONU scheduling and inter-ONU scheduling. Our interest is on how inter-ONU scheduling affects the delay. In another simulation, inter-ONU scheduling is eliminated, all packets are sent with T_s of 100 μs.

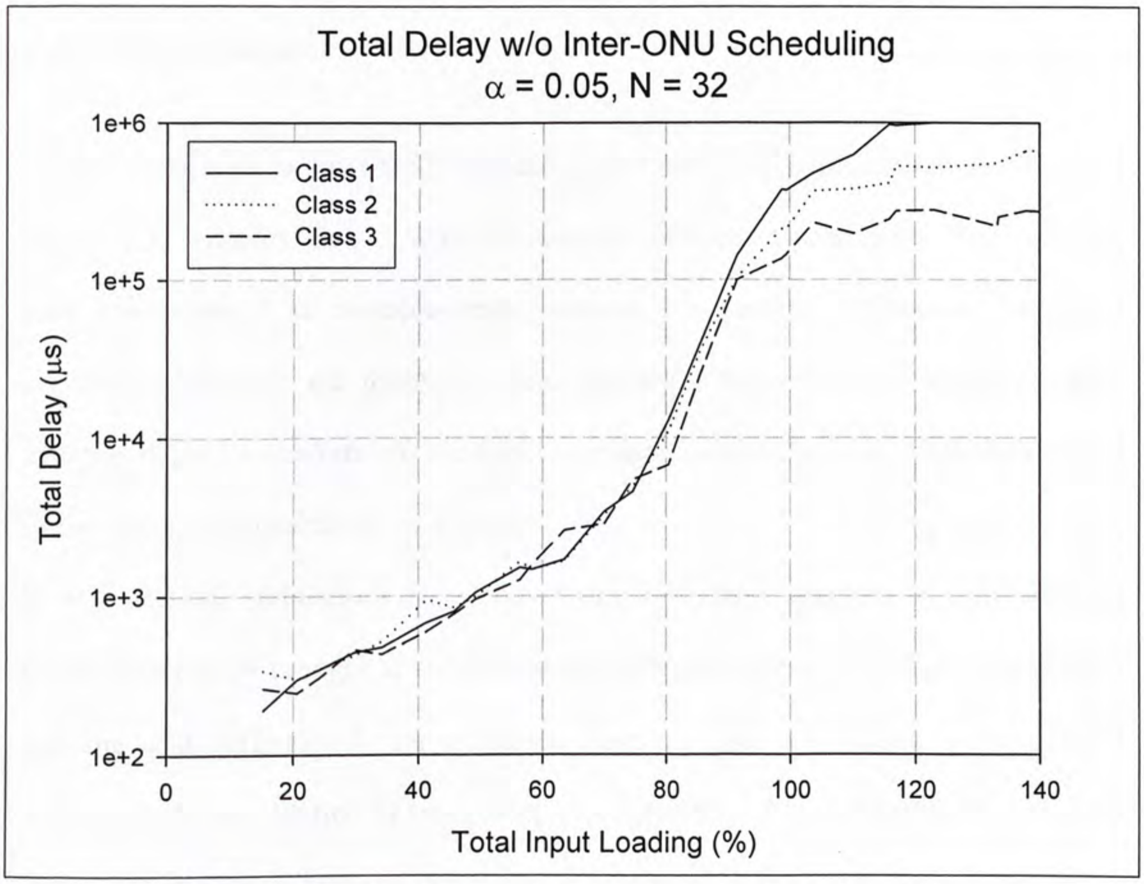

Figure 4.13 Total Delay without using Inter-ONU Scheduling

As we can observe in the 4.13, Class 3 service can no longer maintain small delay. This shows that inter-ONU scheduling plays an important role on supporting CoS. The proposed inter-ONU scheduling provides a possible way to solve the problem by giving ONUs priority.

4.3.5. Conclusions

We have proposed an inter-ONU scheduling that can be implemented on the proposed protocol to support CoS. With the support of CoS, the network that uses the proposed protocol as multiple-access scheme can provide differential classes of services. Although the protocol cannot guarantee the quality of services (QoS), average delay of packets of the highest priority can keep at a small value, and utilization can be guarantee for this class.

In this section, simulations have shown that low delay and nearly zero blocking probability can be brought to the class with the highest priority by the use of priority queuing and different T_s for different classes. The importance of inter-ONU scheduling is also studied by comparing two scenarios with and without it. Results show that the small average delay can no more be maintained without the use of inter-ONU scheduling.

Chapter 5:

Conclusions

Passive Optical Network (PON) access systems have become more mature in the past decade with active research and development. There have been many proposals on the implementation of Optical Network Unit (ONU) and Optical Line Terminal (OLT) with low cost and small size. A problem occurs in upstream access due to its multi-point-to-point (MP2P) nature. IEEE802.3ah and Ethernet in the First Mile Alliance (EFMA) have proposed a solution using a MAC layer protocol for signaling. There are researches on using distributed approach such as CSMA/CD to solve this problem.

5.1. Thesis Summary

This thesis presents an alternative solution for MP2P upstream access in PON. The contributions include:

1. Proposal of CSMA/CA using Pilot Tone on PON with signal feedback to solve the P2MP upstream access problem.
2. Performance evaluation on the proposed algorithm. We compared the proposed scheme with a solution using CSMA/CD scheme. Utilization and delay characteristics are analyzed. We also investigated the effect of network size and packet duration on the performance of the protocol.

3. Utilization Enhancement. We proposed a modification to enhance the utilization performance, with a trade-off of complexity.
4. Capture Effect Solution for using BEB. The disadvantage of using Binary Exponential Backoff (BEB) algorithm for contention resolution is overcome in a protocol-specific way.
5. Implementation of inter-ONU scheduling to support Class of Service. Small delay and utilization guarantee can be given to the class with highest priority. In order to support CoS, we have shown that it is necessary to give higher priority to ONUs which are sending more important packet.

In Chapter 1, we have given an overview on the current status of PON. Two major types of PON, ATM-PON and Ethernet PON, are studied. After the discussion of the P2MP upstream access problem, possible ways of solving the problem are studied. Finally, we conclude that TDM approach is the most suitable candidate at present for multiple-access on PON.

Chapter 2 gives the motivation on proposing the CSMA/CA using pilot tone protocol. MPCP of EPON is studied. It provides a centralized solution where scheduling works are done in the OLT. This protocol requires network synchronization and registration for newly added ONUs. The advantage of this solution is that it can provide a high utilization and low delay. OLT can give higher priority to important packets. Hence, QoS is easier to implement under this policy.

We have also discussed solutions using distributed approach in which scheduling is done by all active ONUs. In distributed approach, upstream signal is fed back to all ONUs on the network. The CSMA/CD solution tries to sense the collision of packet on the network. If collision is detected, the responsible ONUs will take appropriate action to tackle the collision. They either resend the packet after a random period, or drop the collided packet. Synchronization among ONUs and OLT is not necessary. The disadvantage of such solution is that it can only achieve low channel utilization.

With the inspiration from MPCP and CSMA/CD solution, an alternative that takes the advantage of pilot tone is proposed in Chapter 3. The main idea of this solution is to avoid data collision by carrier-sensing. Similar to MPCP, collision is avoided by making request before sending packet. Hence the solution is named "CSMA/CA using Pilot Tone". Simulation works have been done using JAVA SDK. The results show that it has better performance than CSMA/CD approach in terms of utilization and delay. The cost is its complexity of using pilot tone and requesting.

In Chapter 4, attempts to resolve some problems that may happen on the proposed protocol are presented. The first problem is the low channel utilization when using small packet. Modification on the protocol is made based on sending more packets per request. Simulation shows that the utilization performance for a large α value has been greatly improved. Capture effect brings fairness problems under heavy loading. We show that by having a dynamic T_s value the protocol can moderate capture effect. Lastly, by using the same technique, we have shown how to allow priority on the network. This makes the implement of CoS on the protocol feasible.

5.2. Future Work

We have proposed a scheme, with some variants, for resolving upstream access problem in TDM PON using CSMA and pilot tone. Many simulation works with different parameters values and different modification on the protocols are carried out to show the feasibility of the scheme.

For future work, analytical work can be implemented to complete the study. It is worth studying how the backoff algorithm and PON structure affect the protocol analytically.

There are many parameters in the proposed protocol. We have discussed some of them in chapter 3 on how they affect the performance. Furthermore, instead of using BEB, we can use Uniform Backoff, Geometric Backoff, or even using BEB with binary logarithmic arbitration method. In Chapter 4.2 and 4.3, in which capture effect and CoS are studied, different schemes to determine T_s can be explored. In the future, we can fine-tune these parameters by either analytical study on the protocol or exhausted search using simulation.

The upper layer structure also plays an important role on deciding the best system parameters. The simulation results show that an importance protocol parameter, α value, should be keep as small as possible. This implies that packet size should be large. However, large packet size is usually not a good option for upper layer as the cost of aggregating small fundamental packet is high. Adaptive scheme can be made by enabling the use of smaller packet size for light loading to reduce the cost for aggregating, and larger packet size for heavy loading to achieve better system performance.

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